ATEUS ® - DIAL-THRU GSM GATEWAY
Ordering Nr. 501106E

User Manual

2N ®

TELEKOMUNIKACE

Version of manual: 1.0

Version of firmware: 2.50

Dear customer,

Let us congratulate you on purchasing our *ATEUS*® *-* **DIAL-THRU GSM GATEWAY**. During the development and production of this product, care was taken to maximise its value, quality and reliability. We hope you will use the GSM Gateway to it full potential with long lasting benefits.

! Important !

- The manufacturer currently updates the firmware integrated into this product. The ISP technology (In System Programming) allows you to load the latest software versions via the RS232 port on the unit from any computer. For the latest software version including all accessories please refer to www.2n.cz, for instructions please see the **"Control Software Upgrade"** section of this manual. We recommend you use the latest software version in order to guarantee the latest functionality of this GSM Gateway..

- To program your GSM Gateway parameters using a PC you need the "GSM – Program Software". For the most recent version of this programming tool please refer to www.2n.cz.

- You will find the latest version of this Manual in the popular .PDF format on www.2n.cz also. You are recommended to view the latest version in order to find explanations of new functions necessary for any software updates.

- Please read this Manual carefully before installing and using this product for the first time. The manufacturer is not liable for any loss incurred by the user as a result of incorrect usage of the unit. Our warranty terms and conditions do not cover damage caused by rough handling, improper storage, or exceeding the specified technical parameters.

- This Manual is quite comprehensive and includes sections that are not applicable for the basic installation of the unit, other section also include information that may not be applicable to your particular. Please note your gateway model number and refer to these sections only.

- Preliminary information on functions that will be available in later software releases will have a light-grey background or in grey font.

History

Version	What has changed or new in this version
1.0	- Common information taken from the *ATEUS* ® - GSM GATEWAY COMPACT 2000 manual, version 10.0 - New parameters: 701 to 706 - Call Sorting Table expanded to 250 records, resulting in following changes: - Intelligent Incoming Call Routing Table reduced to 50 records - Service buffer disabled

Checklist

Packaging list, please check the contents of your unit:

Item	Quantity	Notes
GSM Gateway 501106E	1	
Mains (A.C. power supply) cord	1	1)
Telephone line cord	2	
Serial cable	1	
Antenna	1	
Holder (for fixing to the wall)	1	
Rawlplugs	2	
Screws	2	
This manual	1	
Warranty Certificate	1	
Software on CD-ROM	• *GSM program* • *SMS program* • *Driver for PC*	

Contents

1. Introduction

1.1. Purpose

- *ATEUS* [®] - DIAL-THRU GSM GATEWAY operates on single analog line between PSTN and PBX (or a telephone set, coin-operated automatic machines, etc.). It allows its users to make calls via the cheaper way, automatically switching between PSTN and GSM network.
- The voice mode, i.e. an outgoing or incoming call, is the basic function of the GSM Gateway. The Gateway is equipped with all functions necessary for this purpose and offers ease of use in this mode.
- Moreover, the GSM Gateway provides (in connection with a PC) data mode and SMS receive/send mode too. These additional functions increase the utility value of the product.
- You need no additional equipment (mains adapter, external GSM telephone) to run the GSM Gateway. The installation is so easy that even a non-professional can install it. All programmable parameters are set at optimum values by default. You need only to fill in the Call Sorting Table for correct outgoing calls routing. Once you have connected the telephone line, antenna, power supply and your SIM card, you can start making calls without hesitation.

1.2. How to Save GSM Call Costs

- By connecting a GSM Gateway between your PBX or phone and PSTN you can make direct calls into a mobile network. **This saves PSTN – GSM connection costs.** You need to fill in the Call Sorting Table to set which prefixes will be redirected to GSM.
- Mobile telephone calls made by your colleagues from outside to your headquarters will be cheaper too, if they will call to GSM number of your gateway.
- With the GSM Gateway you can use **the most convenient tariff rate of your GSM** operator, because calls of all your GSM Gateway users will be billed together.
- If you use **an answering and recording machine** – a GSM service, you may pay for retrieving messages. If you connect an answering machine of your own to the GSM Gateway, **you pay nothing for the retrieval**.
- With the GSM Gateway you can eliminate selected numbers. **You won't pay for a call that is disabled.**

1.3. Other Advantages and Applications

- You can establish a **telephone connection even where there are no fixed telephone lines available** (exhibitions, fairs, conferences, chalets...).
- You are not exposed to the high-frequency electromagnetic field as with a mobile telephone.
- You can also attach a coin-telephone to the GSM Gateway, as it is able to send tariff pulses. You can assess the price for call connections yourself (with profit).

1.4. Main Features

- **DTMF dialling**
- **Pulse dialling**
- **Operates on PBX CO line, between PBX and PSTN**
- **Operates with answering machine or telephone, between this device and PSTN**
- **GSM module SIEMENS** [R]**, 900 and 1800 MHz band**
- **High quality voltage protection at line interfaces**
- **No external mobile phone needed**
- **An easy installation**
- **Call sorting table – call routing to PSTN, GSM**
- **Barring possibility for selected calls**
- **Intelligent end of dialling recognition – faster connection**
- **Intelligent Incoming GSM Call Routing**
- **Tariff pulses: passed thru from PSTN, generated by the pseudo – tariff transmitter during GSM calls**
- **Begin & end of call signalling: passed thru from PSTN, generated during GSM calls**
- **Serial port RS-232C – for connecting to any PC**
- **SMS messages can be received & transmitted by PC**
- **Data mode – can be used as a modem with any PC**
- **Programming by phone (limited)**
- **Programming by PC**
- **Remote programming by PC**

2. Basic Installation Instructions

This chapter describes the basic connection of the GSM Gateway that can be made in a few minutes. All you have to do is to connect an antenna, the power supply cable and telephone lines, insert your SIM card and the GSM Gateway is ready to work.

2.1. Proper Location

- The *ATEUS* ® - DIAL-THRU GSM GATEWAY is a transmitter in principle. You must comply with the local regulations and laws in your country pertinent to usage of mobile phones and transmitters!
- The *ATEUS* ® - DIAL-THRU GSM GATEWAY is designed for vertical mounting. For the required working position see Fig.1.
- The GSM Gateway may be operated in a position other than vertical (on a desk, e.g.) for a short time only – for quick maintenance testing, for example.
- For the acceptable range of operating temperature and humidity refer to the "Technical Parameters".
- The GSM Gateway may not be operated at places exposed to direct solar or heat radiation.
- Exceeding the acceptable operating temperature does not have an immediate impact on the GSM Gateway function, but may result in accelerated ageing (of batteries in particular!) and lower reliability.
- The GSM Gateway is designed for indoor use. It must not be exposed to rain, water, condensed moisture, fog, etc.
- The GSM Gateway must not be exposed to corrosive gas, fumes of acids or solvents, etc., or corrosive liquids, during cover cleaning, for example.
- The GSM Gateway is not intended for use in high-vibration locations such as means of transport, machine rooms, etc.
- The GSM Gateway should be located with respect to the GSM signal quality.
- A free space should be left over and under the GSM Gateway for cables and flowing air that removes heat produced during the operation.
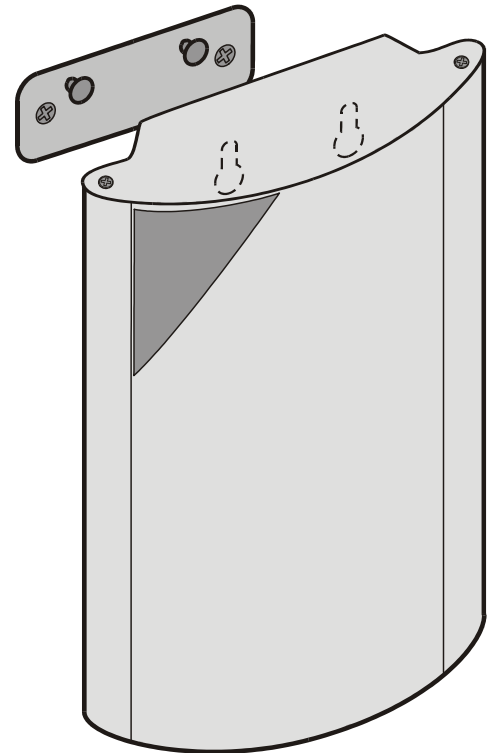
Fig. 1.: *GSM Gateway Working Position*

2.2. Telephone and PSTN Line Connection

2.2.1. Connection to PBX

Connect the *ATEUS ®* - DIAL-THRU GSM GATEWAY to external (C.O.) line of your PBX and to PSTN line socket, as shown on picture. If your PBX has more PSTN lines, make sure that outgoing calls to GSM will go out through this one or use more GSM gateways.

Fig. 2.: *Connection to PBX*

C.O. line preferred by PBX for outgoing calls
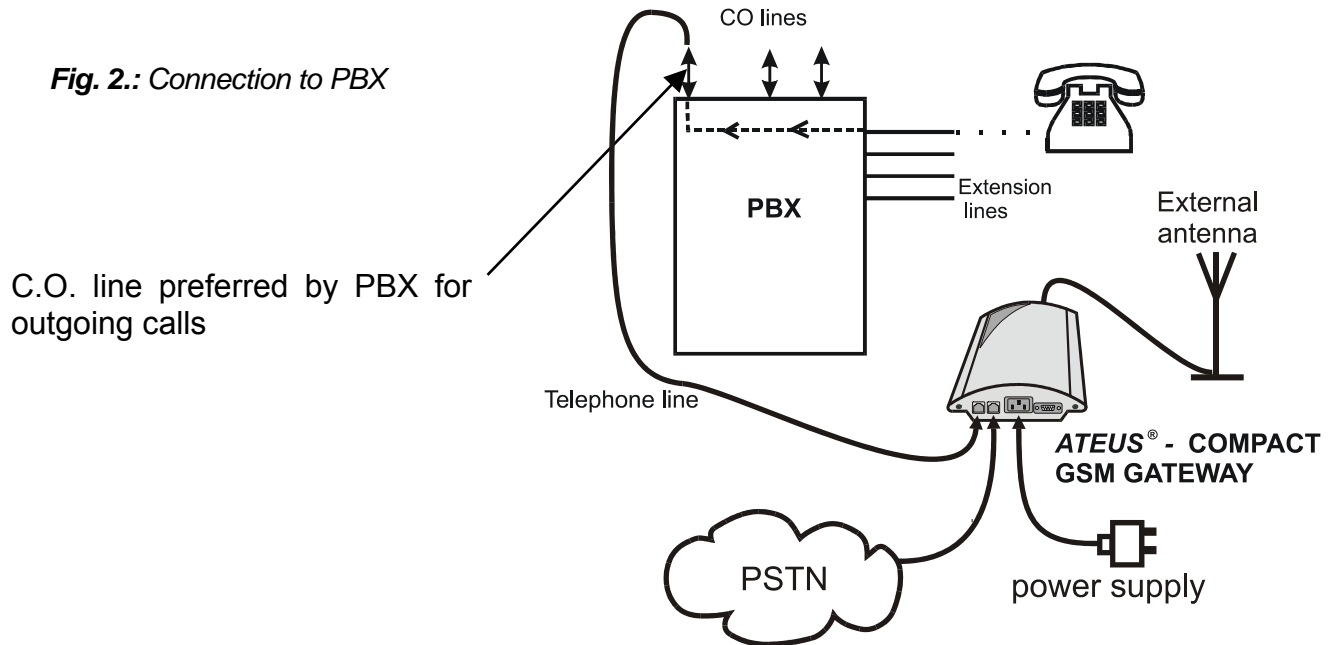
2.2.2. Connection to Telephone Set (Answering Machine, Coin-Phone etc.)

Connect the *ATEUS* $^®$ - DIAL-THRU GSM GATEWAY to PSTN line socket and to your telephone set or some other terminal equipment. For convenience, you can add "***ATEUS*** $^®$ **- Ping pong**" (intelligent double or triple branch made by 2N TELEKOMUNIKACE a.s., order Nos. 831127, 831128, 831137, 831138) to interconnect several devices, such as a telephone set and an answering machine; see Fig. 3.

Note: *If you connect a coin-phone, be sure to program the transmission of tariff pulses and pseudo tariff metering! Remember also that tariff pulses are not generated during PSTN calls. They must be generated by PSTN.*

Fig. 3.: *Optional Connection of More Terminal Equipment*
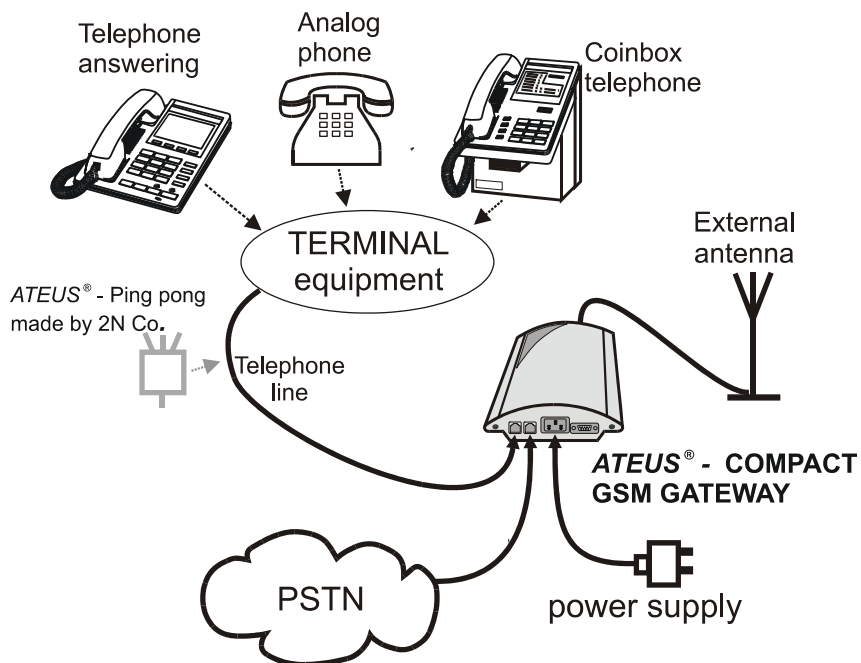
2.3. External Antenna Connection

Connect an antenna or an external antenna cable into the FME connector. The antenna location should have a good GSM signal. The antenna should be in the vertical position. For antenna and cable parameters refer to the "*Technical Parameters*". **Tighten an antenna connector gently by hand; do not use any tools!**

2.4. SIM Card Set-up and Installation

2.4.1. Operator / SIM Card Selection

To perform this GSM Gateway function you need a SIM card of a GSM network operator, using the 900 MHz or 1800 MHz band (depending on GSM gateway model). The *ATEUS* ® - DIAL-THRU GSM GATEWAY works with 3V SIM cards. All SIM cards except for the oldest ones meet this condition. If you are not sure, ask your GSM operator about voltage of your old SIM card. If your SIM card is new or you are going to buy a new one, you need not worry – your SIM card will be O.K.

2.4.2. PIN Entering Blocking (Optionally)

The GSM Gateway provides automatic PIN entering by default. You can disable PIN entering on your SIM card (using a mobile telephone into which you insert your SIM card for this purpose). If you do disable, you need not worry as to whether there is a PIN code stored in your GSM Gateway memory. If you enable PIN entering, your GSM Gateway will require a PIN code after the first power-on and if you enter the PIN correctly, the GSM Gateway will store it in its memory and enter automatically in the future.

2.4.3. GSM Network Service Setting (Answering Machine, Call Forwarding)

Before the SIM card installation decide whether you will use the **incoming call forwarding** service provided by GSM networks (call forwarding in the event of busy line, absence, unavailability...). However, it is more convenient to disable all call forwarding modes (the GSM operator's answering machine, e.g.) and use an answering and recording machine of your own. If you have more GSM Gateways with your PBX, you can forward calls when one GSM Gateway is busy, etc.

2.4.4. Roaming Parameters Setting (Calling via Foreign GSM Networks)

The GSM Gateway disables roaming by default. It is usually convenient because most people do not travel with the GSM Gateway and there is a risk with roaming in foreign countries that, due to a failure in the local GSM network, you might get registered in another network and pay much more for your calls. To enable roaming and set network preferences, complete the list of GSM networks to be preferred using your mobile telephone and then enable roaming while programming the GSM Gateway.

The registration of the GSM Gateway in a foreign GSM network is signalled by a special dial tone ─ ─ ── (refer to the list of tones) and you have to dial numbers including international prefixes that can be easily barred (refer to *Programming, Call Sorting Table*).

2.4.5. SIM Card Insertion in GSM Gateway

To install the SIM card, press the yellow button on the SIM holder using a suitable tool (e.g. scissor) to make the drawer slide out. Pull out the drawer, insert the SIM card in it, slide the drawer back and click into position.

2.5. Power Supply Connection

- Make sure that the voltage in your mains corresponds with the data on the product label.
- Make sure that the antenna has been connected. **If you connect a power supply to the equipment without the antenna, you might cause damage to the GSM module transmitter.**
- Connect your power cord. After a while, the green indicator "AC Supply O.K." should go on.

2.6. Functional verification

Recommendation: check GSM gateway functionality prior to connecting it to your PBX and programming.

1. Connect GSM gateway to previously checked telephone set and PSTN line. Verify that this telephone is switched to DTMF and its ringer is ON. It is better to check a SIM card too, using a mobile phone. It is not necessary to fix GSM gateway to a wall for testing – it can lie on flat surface as well. Excepting this, follow chapters 2.2 to 2.5 to connect all needed.
2. If the inserted SIM card requires a PIN, the red lamp "Enter PIN" will light up. In this case, pick up the handset of connected a phone. You will hear a PIN tone ---- ---- ----. Enter PIN as described in chapter 3.9.2 and hang up, the red "Enter PIN" lamp will go out.
3. The GSM gateway will register itself into the GSM network. First, the red "No GSM network" lamp must go out. After a moment, the green "GSM ready" lamp will light up.
4. Pick up the phone; you will hear the dialling tone ─ ── and the "Line ready" lamp will start blinking. If it doesn't, the phone or its connection is bad.
5. Now check for signal quality. Enter programming mode according to chapter 6.3.2, skipping step No. 1 (this applies only in the case of connection to a PBX). Indication of GSM signal quality will be turned on automatically. As more lamps are lit, signal quality is better. If at least one green lamp is lit up, signal quality is excellent. Try to find a good place for the antenna. Keep the antenna vertical and move it slowly – signal quality information is updated every three seconds. Remember that a movement as small as 10 cm may have a considerable effect on signal quality, as well as a position close to your body. The best way is to step aside after each relocating of antenna. Hang up after positioning the antenna; do not program anything!
6. Make an outgoing call. Remember that all calls will be routed to GSM by the factory default. If your GSM gateway is not used first time and the Call Sorting Table is not empty, choose a number, which will be routed to GSM. Call your colleagues mobile e.g. and verify that you hear each other well. In the case of a completely new pre-paid SIM card, one outgoing call is necessary for SIM card activation. Until it is activated an incoming call cannot be received! Make an incoming call now. Call the GSM gateway from some mobile; the phone should ring.
7. To check PSTN interface, use an incoming call. Ask somebody to call from some phone to PSTN line which is connected to your GSM gateway; the phone should ring. Pick it up and verify, that you hear each other well.
8. The GSM gateway is now checked. You can now connect it to a C.O. line of the PBX. After connection is completed, fill in the Call Sorting Table (for programming see chapter 6). Check incoming call and outgoing calls again, both PSTN and GSM. This may, of course, necessitate some programming or settings changes of the PBX. If everything is O.K., you can go to programming, if it is required - see chapter 6.

2.7. LED Indicators

color, name	Description of statuses
green POWER	• lights = GSM module is powered • blinking slowly = GSM module is not powered (c. 6 seconds after switching on)
green GSM	• lights = registered into native GSM network • blinking = registered into GSM, roaming • dark = not registered into GSM sítě
yellow LINE	• lights = GSM call (after connect) • blinking slowly = PSTN call (after dialling) • blinking fast = off hook before call, making connection, incoming ringing • dark = on-hook, or line error
yellow DATA	• lights = GSM data connection established • blinking = data exchange with PC, GSM data connection not established • dark = no activity on serial interface

red
ERROR
	• lights = at least one from these errors:
	   o line error
	   o SIM is not present
	   o PIN is not entered
	• blinking = GSM signal quality indication
	• dark = no error
	(ATTENTION! It doesn't imply, that GSM gateway is registered into GSM network!
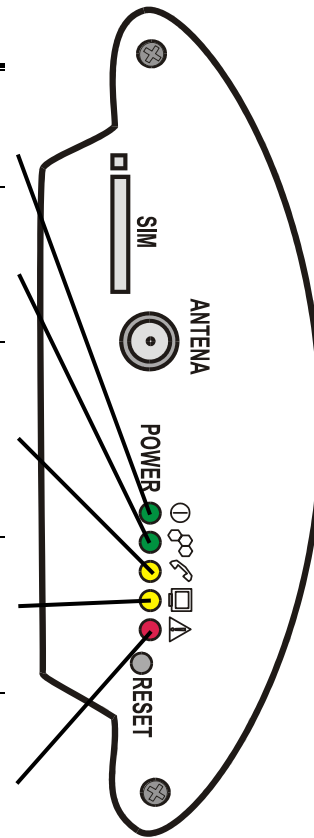
3. User Manual – Description of Basic (Voice) Function

Users mostly use their PBXs and GSM Gateways intuitively, without reading any instructions, or follow very simple instructions provided by an authorized person. The following functional description is therefore intended for technicians, who follow the instructions (depending on the PBX set-up) and solve any operational problems.

3.1. Outgoing Call to GSM

3.1.1. Picking Up the Line

The PBX picks up a line the moment a subscriber picks up his or her handset and dials a number (or prefix) that is routed outside.

Note: If the GSM Gateway is busy, the PBX can either give the caller the busy tone — — — — — or choose another connection (PSTN, or there may be more GSM Gateways with one PBX).

3.1.2. GSM Gateway Ready Signalling

The GSM Gateway registers the off-hook (current inflow). Immediately and then, if everything is O.K., starts sending its usual dialling tone − —. Now the subscriber can dial the number.

Notes:

- If GSM gateway needs PIN, special tone is transmitted. See chapter 3.9. Until the correct PIN is entered, GSM gateway will not allow any outgoing call. Only an incoming call from PSTN is allowed.
- In some cases, PBX operates as a repeater. Subscriber will dial whole number, then PBX repeats it to PSTN line. In this case, subscriber will not hear dialling tone from GSM gateway.
- *As far as pulse dialling is selected, DTMF dialling can be used to enter programming mode and program GSM gateway by phone. In case of any other DTMF dialling, GSM gateway will answer by busy tone.*
- If one network is not accessible, an outgoing call to second one is still possible. User will hear normal dialling tone, but dialling may cause a busy tone - if inaccessible network is required.
  GSM network is not accessible in these instances:
  - GSM module in use (data mode)
  - No SIM
  - GSM network failure, no antenna etc.
  PSTN network is not accessible in the case of PSTN line failure - no current.
- In this moment, GSM gateway also picks up the PSTN line. This help to avoid a conflict with incoming call from PSTN.

3.1.3. Dialling Receive

The GSM Gateway is ready to receive pulse or tone dialling (according to the set-up). As soon as the subscriber starts dialling a number, the GSM Gateway mutes the dialling tone − — (as with public telephone exchanges). The user must dial digits in no more than 6 second intervals; otherwise the number is regarded as complete and sent to the GSM network (this timeout is programmable).

Notes:

- **If pulse dialling is selected, and the call is routed to PSTN, dialled number is transformed to DTMF.**
- *If pulse dialling is selected, DTMF can be still used for programming.*
- *Some PBXs analyse the whole number first and then transmit the dialling into the CO line (GSM Gateway). Here, the signalling type and the timeout depend on the PBX set-up!*

3.1.4. Dialling End Recognition

The GSM Gateway itself can recognize the end of some numbers according to their length. Moreover, you can set your GSM Gateway in such a way that the GSM Gateway accepts the '*' or '#' (for tone dialling only) symbols as the end of dialling. Otherwise, it waits 6 seconds after the subscriber stops dialling (the timeout is programmable). Then, the subscriber can hear a short tone − signalling the dialling end.

Note: *If the caller goes on dialling, the GSM Gateway will not accept the extra digits!*

3.1.5. Call Routing

According to a record found in the Call Sorting Table, call is routed to GSM or PSTN network or bared. If the required network is not accessible, GSM gateway starts to send busy tone or some special tone (PIN required etc.). Otherwise, the GSM Gateway transmits the received number into the right network. Description for the case of GSM follows.

3.1.6. Connection Making and Establishing

In this moment, GSM gateway sends whole received number (or a number changed by "take away" and "append" parameters) to the GSM network. Next, GSM network is making a connection, and it takes typically 8 seconds. During this time, the subscriber hears a special "call progress" tone (differs by GSM gateway model and version of software). Next, the subscriber usually hears the ringing tone ⎯⎯ ⎯⎯ or another signal transmitted by the GSM network. The connection, however, is not established and paid for until the called party answers the phone. The GSM network signals this moment and the GSM Gateway can pass the information to the PBX. If this type of signalling is used (exceptionally), the calling party can hear a click in the earphone.

3.1.7. The Call

The call may be terminated forcibly if the GSM signal gets lost, for example, or in similar situations.

During the call, GSM gateway keeps PSTN line off-hook. This is important because GSM gateway is not able to serve incoming call from PSTN, till PBX line is busy by outgoing GSM call. If somebody will call to you (via PSTN line), he or she will get busy tone from public exchange.

3.1.8. Tariff pulses

During outgoing call to GSM, the GSM Gateway can transmit pseudo-tariff pulses, according to the Call Sorting Table.

ATTENTION! *If filled wrong, it may cause that these pulses will not be transmitted, or will not show a true price of the call.*

3.1.9. Connection Termination (End)

If the caller is the first to hang up, the GSM Gateway registers the on-hook immediately (the current flow stops) and terminates the connection. If the called party is the first to hang up, the GSM Gateway gets the information from the GSM network and terminates the connection. The GSM Gateway can pass the information to the PBX. The calling party gets the busy tone − − − − − (or another type depending on the set-up).

The time of call may be limited by parameter 158. 30 sec before this limit, GSM gateway sends a warning tone. Last 10 sec a short beep repeats each second. A call interruption follows, optionally busy tone and Power Down.

Note: *With some calls, the called party's on-hook information is considerably delayed by GSM network (30s, e.g.). The subscriber usually registers the on-hook earlier, hangs up, and the GSM Gateway terminates the connection immediately.*

3.1.10. Subscriber's Disconnection (Power Down)

If a subscriber blocks the GSM Gateway by seizing the line without dialling a number, or fails to hang up after the call, he or she will get the busy tone − − − − − first and then is disconnected (Power Down status).

3.2. Outgoing Call to PSTN

Till call routing, GSM gateway behaviour is the same as during GSM call. See chapter 3.1.1 to 3.1.4 for its description.

3.2.5. Call Routing

According to a record found in the Call Sorting Table, call is routed to GSM or PSTN network or bared. If the required network is not accessible, GSM gateway starts to send busy tone or some special tone (PIN required etc.). Otherwise, the GSM Gateway transmits the received number into the right network. Description for the case of **PSTN** follows.

3.2.6. Connection Making and Establishing

Remember that the PSTN line is already off hook (see chapter 3.1.2, last note). In this moment, GSM gateway checks how long is PSTN line off hook. If parameter 701 is over, GSM gateway hangs up for a moment, using parameter 702. (This happens when a dialling was very slow.) Next, whole number is repeated to the PSTN network (or a number changed by "take away" and "append" parameters). DTMF is used always, even if pulse dialling was received. Next, PSTN network is making a connection. During this time, the subscriber may hear tones from PSTN. Next, the subscriber usually hears the ringing tone ———   ——— or another signal transmitted by PSTN network.

3.2.7. The Call

During the PSTN call, incoming calls from GSM network are refused. (This is important because GSM gateway is not able to serve two calls at a time. Without refuse, calling party hears ringing, but it is not possible to ring to PBX.)

3.2.8. Tariff pulses

During the outgoing call to PSTN, tariff pulses, if present, are passed from PSTN to PBX. Pseudo-tariff parameters in the Call Sorting Table are ignored.

3.2.9. Connection Termination (End)

GSM Gateway registers the on-hook (the current flow stops) and switches to its default state. It is important to secure that PBX will always hang up the line. The time of PSTN call is not limited by parameter 158.

3.2.10. Subscriber's Disconnection (Power Down)

If a subscriber blocks the GSM Gateway by seizing the line without dialling a number, or fails to hang up after the call, he or she will get the busy tone − − − − − first and then is disconnected (Power Down status).

3.3. Incoming Call

3.3.1. GSM Gateway Ringing, Extension Dialling, Extension Ringing and Connection Establishing

When the GSM Gateway receives a command from the GSM network and, if available, the CLIP information, it starts ringing (i.e. generating the ringing voltage – whose timing is programmable) into the PBX. The PBX registers the ringing and then, one of the following situations may occur:

3.3.1.1 PBX without DISA = Selected Extension Ringing

In this case, the selected extension (or several extensions at the same time or sequentially according to the PBX set-up) starts ringing and the calling subscriber will not pay for the call until the ringing extension answers.

3.3.1.2 PBX with DISA, Intelligent Routing Off

In this case, the PBX answers and starts reproducing the so-called DISA message. The GSM Gateway establishes connection immediately in order that the caller can hear the message and dial the required extension.

3.3.1.3 PBX with DISA, Intelligent Routing On, and CLIP Present and Known
(Only for GSM calls. GSM gateway cannot receive CLIP from PSTN line.)

In this case, the PBX also answers and starts reproducing the DISA message. The GSM Gateway, however, has found the caller's number in its Intelligent Incoming Call Routing Table and thus knows the extension to be called. Therefore, the GSM Gateway does not establish connection immediately, but serves the DISA function (waits and dials the extension number). Then, it establishes the connection and the calling subscriber can hear the ringing tone ⸺ ⸺ and the called subscriber.

3.3.1.4 PBX with DISA, Intelligent Routing On, but CLIP Absent or Unknown

In this case, the PBX also answers and starts reproducing the DISA message. The GSM Gateway, however, has not found the caller's number in its Intelligent Incoming Call Routing Table (or has not received the CLIP). Then it can (according to its set-up) either work as described in 3.3.1.2, or as described in 3.3.1.3, plus dial the operator's number.

3.3.2. Call

With incoming calls, the GSM Gateway waits until the call is terminated, which situation is the same as with an outgoing call.

Another differences between GSM and PSTN calls:

3.3.2.1 GSM incoming call

GSM call may be terminated forcibly if the GSM signal gets lost, for example, or in similar situations.

During the GSM call, GSM gateway keeps PSTN line off-hook. This is important because GSM gateway is not able to serve incoming call from PSTN, till PBX line is busy by GSM call. If somebody will call to you (via PSTN line), he or she will get busy tone from public exchange.

3.3.2.2 PSTN incoming call

During the PSTN call, incoming calls from GSM network are refused. (This is important because GSM gateway is not able to serve two calls at a time. Without refuse, calling party hears ringing, but it is not possible to ring to PBX.)

3.3.3. Connection Termination (End)

For GSM call termination, see chapter 3.1.9
For PSTN call termination, see chapter 3.2.9

3.3.4. Subscriber Disconnection (Power Down)

If a subscriber blocks the GSM Gateway unnecessarily by not hanging up after the call, he or she will get the busy tone − − − − − first and then is disconnected (Power Down status).

3.4. Begin and end of connection signalling

Signalling by a current break or polarity change (see parameters 231 to 234) works differently in a case of GSM and PSTN call:
GSM call: signals may be generated by GSM gateway, according to parameters 231 to 234.
PSTN call: signals may be only passed from PSTN network.
NOTES:
- *This difference may cause some problems, be careful when using this function.*
- *GSM gateway is physically switching PSTN line to PBX line during PSTN calls. To avoid possible polarity change in the moment of switching, GSM gateway checks PSTN line polarity after power on and optionally reverse polarity of PBX line to match polarity of PSTN line. When connecting to other PSTN line, always switch GSM Gateway off. Otherwise, polarity change may cause some problems, and also audible clicks are louder.*

3.5. Power failure

During power failure (or if power cord is disconnected), PBX and PSTN connectors are interconnected. It means that all calls are realized via PSTN, and GSM gateway cannot bare any calls, of course.

3.6. Automatic dialling ("baby call")

Outgoing calls are described in chapters 3.1 and 3.2 on the assumption that automatic dialling is off (default). If parameter "Number for automatic dialling" is filled, this function is automatically switched on and GSM gateway operates as follows:

As soon as line is picked up, GSM gateway awaits dialling for limited time, defined by parameter "time for automatic dialling". If user starts dialling within this time, GSM gateway operates normally, as described in chapter 3.1. Otherwise, if this time is out, GSM gateway automatically makes a call to programmed number. "Baby call" is always routed via GSM network.

Note: It is assumed, that GSM gateway is connected to phone. If GSM gateway is connected to PBX, applicability *of automatic dialling function depends on PBX's settings.*

3.7. Intelligent Incoming Call Routing

This function can be applied to incoming calls from GSM only (if enabled). If CLIP (number of calling subscriber) is found in Intelligent Incoming Call Routing Table (see chapter 7.4.), GSM gateway will call a dedicated extension according to this table. If GSM gateway is connected to external (C.O.) line of PBX, DISA function must be used in PBX to serve it. This function is operating automatically, as described in previous chapter 3.3.

3.7.1. Intelligent Incoming Call Routing Control

From the viewpoint of the user, this function can work completely automatically but moreover it is possible to complete the Intelligent Incoming Call Routing Table during any call by special command:

Command for Intelligent Incoming Call Routing fill-up:

Example: 7 7 7 ⊛ 2 3 4 ⊛   (route this subscriber to extension 234)

Extension number entering, or #️ for command cancelling.

Zero instead of extension number - enables dialling to calling party

Extension number, up to 8 digits

Empty - erases a record with this CLIP, if found.

Password entering, after it GSM gateway will send confirmation signal. Or #️ for command cancelling.

Password for fill-up into table

If the GSM gateway executes this command, it will send a confirmation signal as well as sending an error signal in the following events:

- Routing is disabled (error will be sent as soon as password is entered).
- Whole table is "locked" – only programming by PC can modify it.
- CLIP is unknown (error will be sent as soon as password is entered).
- CLIP is already stored in locked part of table; this entry has a priority and cannot be modified.

Notes:

- *If 0 (zero) is entered instead of extension number, GSM gateway enables calling party to dial any number. This is a way, how to enable this possibility only to selected persons. Another incoming calls are connected to pre-selected extensions or refused (dependent on a GSM gateway configuration).*
- *Only <u>unlocked part of table</u> can be filled-up by this command. Once it is full, new ones will overwrite oldest entries. Size of available unlocked part is from 0 to 99 entries, depending on size of locked part. Only programming by PC can modify the size of locked part and its content.*
- *This command is ignored until a connection is established.*
- *In the case of outgoing calls, the called subscriber number is usually incomplete (without international prefix). In case of incoming call, CLIP is complete and international prefix starts with "+". To make these numbers comparable, incomplete number is completed automatically within writing to the table, this way:*
  - *If called number begins with "00" (or with different international prefix defined by parameter 115), it is removed and only "+" character is added to its beginning.*
  - *If called number begins with one "0" (or with different long distance code defined by parameter 117), it is removed and "+" and your country code is added to its beginning.*
  - *In other cases, "+" and your country code is added to its beginning.*
- *While programming by PC, an incomplete CLIP can be entered – e.g. bare international prefix. In this example, incoming calls from each country will be routed to the person who is proficient in the appropriate language etc.*
- *While programming by PC, each CLIP must begin with country code.*

3.8. Telephone Line Tones, Ringing Course - Summary

The *ATEUS* ® - DIAL-THRU GSM GATEWAY transmits tones to the telephone line that signal its operating status. The frequency is 425 Hz for all tones.

Common Dial tone: ─ ──
- The equipment is registered in the domestic GSM network.
- The equipment is ready to receive dialling.
- This tone has the same parameters as the PSTN dial tone.
- The parameters of this tone are programmable.

Special Dial tone: ─ ─ ──
- The equipment is registered in a foreign GSM network – ROAMING.
- The equipment is ready to receive dialling.
- The parameters of this tone are programmable.

Ringing Tone: ─── ───
- The called subscriber is free and his or her telephone is ringing.
- The GSM network transmits this tone; its parameters are beyond the control of the GSM Gateway.
- During an outgoing call to PSTN, this tone is transmitted by PSTN network.

Busy Tone: ─ ─ ─ ─ ─
- This tone is transmitted if:
    - The SIM card has not been installed.
    - The GSM Gateway is not registered in the GSM network.
    - The equipment is registered in a foreign network, but roaming is disabled.
    - The called number has too many digits (over 30).
    - The called subscriber is busy.
    - The called number is bared by call sorting table.
    - The connection has been terminated.
    - GSM Gateway is in data mode.
    - There is a communication error between the control processor and the GSM module, and a servicing intervention is required.
- This tone has the same parameters as the PSTN busy tone.
- The parameters of this tone are programmable.

Dialling End Signalling: ─
- The dialling reception is terminated, and the connection is being established.
- 1 tone, 200 ms (programmable).

PIN Tone: ---- ---- ----
- Your PIN code is required.
- Transmitted upon power-on if the PIN code has to be entered manually.

PUK Tone: -------- --------
- Your PUK code is required.
- Transmitted upon repeated incorrect PIN code entering and the subsequent SIM card blocking.

PIN/PUK OK: ─────
- This 2 s long tone signals that the PIN or PUK code was entered correctly.

Ringing Course: ─── ───
The ringing course (1 s ringing, 4 s pause) is the same as in the PSTN, but can be re-programmed any time.

3.9. PIN/PUK Code Entering

3.9.1. Three Ways of PIN Code Entering

With a common mobile telephone, you have to enter your PIN code after power-on in order to be protected against misappropriation (of your powered-off telephone) and misuse. With the GSM Gateway, this situation may occur after power failure. The difference is that there is often no one to know and enter the PIN code after power recovery. There are three ways in which to solve this situation:

a) **Enable the SIM card function without PIN code entering:**
This is the simplest solution, but the SIM card can be easily misused when stolen.

b) **Set the automatic PIN code entering:**
The PIN code is entered during programming or after power up of the GSM Gateway as mentioned below and stored in the memory. The PIN code is then entered automatically after every power-on.

c) **Set the manual PIN code entering:**
This is the safest way, which requires manual entering of the PIN code after every power-on. Therefore, it is useful for backed-up models only where such situations are rare.

3.9.2. PIN, PUK Manual Entering

If the PIN ---- ---- ---- or PUK -------- -------- tone is transmitted after picking up the line, enter the required code using the DTMF and verify the dialling with the ✳ key.

Example:

PIN Entering: 5 5 8 9 ✳

PUK Entering: 1 2 3 4 5 6 7 8 3 3 3 3 ✳

Your PUK   New PIN

If you enter the correct code, you will hear a 2 s long tone ————— . If not, the PIN/PUK tone will go on. An incorrect entering (incorrect PIN or PUK, incorrect number of digits, unacceptable characters) makes the PIN ---- ---- ---- or PUK -------- -------- tone being transmitted repeatedly. To delete an incorrect code, press '#' or hang up (before entering '*', of course).

Notes:

- *A four-digit PIN code is used in the example above. An eight-digit PIN code is used exceptionally. The GSM Gateway supports this PIN too, but has no information on how long the PIN should be. Therefore, it transmits the same PIN tone ---- ---- ---- for this PIN code too.*

- ***The GSM Gateway does not support emergency calling without PIN code!***

⚠️ | **WARNING!!!**
You have a limited number of attempts for PIN and PUK code entering.
Any repeated error in PUK entering may cause damage to the SIM card!

3.9.3. Protection against Exhausting All PIN Entering Attempts by Automatic PIN Entering

Every SIM card provides a limited number of PIN and PUK entering attempts. To avoid exhausting of all PIN-entering attempts, as a result of repeated GSM Gateway power on/off after SIM card replacement, for example, ***the automatic PIN entering is disabled temporarily*** in case the SIM card refuses the PIN stored in the GSM Gateway memory. If the PIN is entered manually and is correct, it is stored and the automatic entering is recovered.

3.10. Notes

- **Telephone Line Power Down (Model for External Line of PBX Only)**

  Dialling — ——, busy — — — — —, PIN ---- ---- ---- and PUK -------- -------- tones are transmitted into a line for 60s. When this time elapses, the line is put in the Power Down status (no power supply) until it is hung up. In the programming mode, the line is put in the Power Down after 180s.

- **DISA**

  The DISA service relates to incoming calls only. The GSM Gateway itself is not equipped with the DISA function because it is useless – it is more convenient to use the PBX DISA. For more details on the function refer to par. 3.3.1– *"GSM Gateway Ringing, Extension Dialling, Extension Ringing and Connection Establishing".* If DISA is used, you are recommended to forward incoming calls at night, during absence or busy line to the operator, mailbox or answering machine, because any connection attempt is billed to the calling subscriber. Further, remember that the GSM operator usually limits the ringing time (for 30s, e.g.) and there is not much time for sequential ringing of several lines.

- **"Incognito" (Only for GSM network)**

  This function (refer to the "*Programming*" chapter) prevents the called subscriber from seeing the number of your GSM Gateway. This function can be used, for example, if you want to reduce incoming calls in such cases as:
  - Incoming calls from strangers represent no saving for you, but block your GSM Gateway for your outgoing calls that can save your telephone costs considerably.
  - The subscriber you called (even unsuccessfully) from your GSM Gateway has your GSM Gateway number in his mobile phone without knowing that it is a GSM Gateway number. When calling back, he or she may get through to another person (operator, e.g.) and has to try to get to the person who made the call, paying for all this.

- **"Outgoing Calls Only"**

  This function allows you to refuse all incoming calls. You can use it, for example, when your GSM Gateway is busy making outgoing calls but you do not want to use the Incognito function.

- **GSM Gateway Indicators**

  GSM Gateway indicators are not necessary for every-day operation. They are used for control purposes and indicate most operational statuses and failures. Common statuses are green, less common statuses yellow, and failures are red. Every indicator is provided with a clear text. For details refer to the "*Installation*" chapter.

3.11. Instructions for Use for Common Users

As previously mentioned, subscribers usually use their PBX and GSM Gateway intuitively without reading any instructions, or follow very simple instructions provided by an authorized person. These instructions may differ in details according to the PBX set-up.

You can complete and copy the "aid" included below for all users:

Instructions for GSM Gateway Use

GSM Gateway Calling:

- Dial **..........** before the number.
- If you will hear the busy tone ‒ ‒ ‒ ‒ ‒, try later.
- If the GSM Gateway is ready, you will hear the dial tone ‒ ⸺, dial the number – see below.
- **If you hear another tone, do not dial a number and hang up!**

Number Dialling:

- **Timeout:** If you can hear the dial tone ‒ ⸺, start dialling within .... seconds at the latest!
- **Dialling speed:** Do not make pauses longer than... seconds in the dialling!
- **Dialling end:** When you can hear a short beep, do not go on dialling!
- **Connection acceleration:** if you call a number starting with... you can press... after the last digit to accelerate the connection by several seconds.

Barred Numbers:

Never dial the following numbers; they are barred:

....

Intelligent Incoming Call Routing Command

.... ⊠ Your extension number ⊠

Using this command during a call, you make a rule to forward subscriber, currently talking with you, to the extension specified by you from this time forth.

---

4. User Manual – Description of Data Functions

4.1. Use of Data Mode

4.1.1. Destination:

- For data transfer between two computers (second one can have whichever modem)
- For connecting to Internet
- High speed data (GPRS) mode can be used for connecting to Internet and similar applications (model 501105)

4.1.2. Serial interface

Serial interface connector is D-Sub 9 pins, female, see fig. 9. It is connected like a common external modem. All handshake signals are used in data mode. Bit rate is fixed, see table bellow. All applications must be set to this speed.

Bit rate and data format on the serial interface:

model	Bit rate	Data format
501105 (GPRS)	**57600** bit/s	8 bits, no parity, 1 stop bit (8N1)
All other models	**19200** bit/s	

Note: Bit rate of the serial interface is always higher than data rate from/to GSM network. Please, do not interchange these parameters!

4.1.3. Data Rate in CSD mode

- **Maximum data rate** in this mode is 9600 or 14400 bit/sec. Real value depends on GSM network, its load and signal quality.
- Fax transmitting and receiving is technically possible, but it is not supported by the current version of software yet.
- High-speed HSCSD mode is not supported.

4.1.4. Data Rate in GPRS mode (only for model 501105 now)

Multislot Class 8 (or 4+1, i.e. 4 Rx + 1 Tx) is able to use up to 4 timeslots for download and 1 timeslot for upload. Maximum theoretical data rate for download is 57600 bis/sec. Real value varies during connection, and depends on GSM network, its load and signal quality.

This paragraph is valid only for model 501105. For all other models: GPRS is not supported.

4.1.5. How to combine different modes of serial port

Serial port is used by these applications:
- GSM program
- SMS program
- Your Internet browser, if you choose connection to Internet via GSM gateway
- Your Z-modem or another program, if you connect GSM gateway to another computer

Basic rule: All applications listed above are excluding each other. It means if you need to run another one you must terminate current one first. E.g., if you have SMS program running permanently, and you wish to connect to Internet via GSM gateway, you must terminate it.

4.1.6. How switching between voice and data modes works

If enabled, data mode has same priority as voice mode. Once occupied by a voice (phone) connection, GSM gateway is not ready to handle data as long as this call continues, and vice versa: once occupied by a data connection, GSM gateway is not ready to make a call as long as this data connection lasts.

Important note: current version of firmware for GPRS model is not able to handle any voice calls during all time of GPRS connection (between ATD to ATH commands).

Notes:
- *SMS can be transmitted and received during call.*
- *SMS program reads all new SMS's stored on SIM card right after start. SMS program automatically erases SMS's from SIM card, if you don't disable it. It is adding all new SMS's to file on your PC. Almost unlimited number of SMS's can be stored and viewed this way.*
- *Received SMS commands destined for switch control are erased just after execution. They are recognized automatically and not forwarded to SMS program.*
- *If any SMS's excluding these for switch control will come during time, when SMS program is not running, it can fill SIM card memory. If it will be full and next one will come, an oldest one will be erased in order to not block path for switch control commands. If switches are not used, erasing can be disabled by parameter 109, see chapter 7.1*

4.1.7. Installation of driver on your PC

Driver must be installed only if you are using the GSM gateway as a modem – for data transfer between two PC's or for connecting to Internet. It is not used for SMS program and GSM program.

Install driver from enclosed media (floppy or CD) as a conventional modem driver. Choose **Phone and Modem Options** in folder **Control Panel**, and then choose **General** and **Add**. Next browse a path to driver etc. After it, choose driver version:

Driver versions:

ATEUS GSM GATEWAY (analog 19200bps)	Driver for all models excluding 101105 (GPRS)
ATEUS GSM GATEWAY (analog 57600bps)	Driver for GPRS model, CSD mode *)
ATEUS GSM GATEWAY (GPRS 57600bps)	Driver for GPRS model, GPRS mode *)

*) Currently not available on DIAL-THRU model

Important notes:
- It is not recommended that you use the original driver for TC35, available on SIEMENS® web site. This driver allows setting some parameters, which may cause fatal problems within voice connection.
- In folder **Diagnostics** button **Diagnostics** is not working (on both versions of driver)

4.1.8. How to adjust connection to Internet

If you have the driver (see above) and Internet browser installed, it remains only to **make new Connection**.
First, choose GSM modem. In case of GPRS model (501105) you can choose CSD or GPRS connection, see table above. GPRS connection is recommended (check if operator and used SIM card supports this service).
Next, read all the instructions of your GSM network operator – settings are very different! Usually you will find detailed step-by-step instructions on your operator's web site. It is mainly a number to be dialled and other instructions.
In case of GPRS connection, APN setting is required. Follow GSM operator instructions. APN is set by command "Extra settings" in folder "Advanced Connection Settings".

Example: at+cgdcont=1,"IP","internet.click.cz"

Attention: For GPRS connection, operator's instructions may contain telephone number like *99#. In this case, use number *99***1#. Added characters sets PDP context 1, which is necessary in order to set APN by "Extra settings" *command (see above).*

4.1.9. Serial port functions – for experts

- If GSM gateway is ready, it will send back all commands (echo)
- Connection is signalled by DCD output
- Incoming data is signalled by RING output and GSM gateway will transmit: +CRING: <type>+CLIP:"<clip>", 145
- It is possible to select after how many rings the GSM gateway should answer an incoming data call automatically. You can set it by parameter 181 or the ATS0 command. If you used the ATS0 command, entered value remains until the GSM gateway is powered up or reset, or until another change by the AT command. After the GSM gateway power up or reset, the function value is set according to parameter 181. The ATZ and AT&F commands set the function value according to parameter 181 too. This function has no effect upon incoming voice calls.
- Incoming voice call is not signalled by RING.
- Multiple AT-commands are not supported.
- SMS's can be handled directly by supported set of AT-commands, including time of voice call.
- During a voice call, all other AT-commands GSM gateway refuses by BUSY or ERROR.

4.1.10. Supported AT-commands

These commands are available on the Internet, www.2n.cz.

4.2. PC-Based SMS Receive/Send

This program works like common e-mailing software, under Microsoft® Windows® 95 and higher. It can receive, store, edit and send SMS's on the PC, connected to the GSM gateway by serial cable. This program is freeware and the actual version is available on the Internet, www.2n.cz.

5. Installation Instructions for Advanced Users

⚠️ **ATTENTION! DANGER!**
Draw out an AC mains cord before opening a cover!
Risk of an electric shock!

⚠️ **WARNING!**
All removable parts of cover are **earthed** with earthing cables!
We do not recommend that you disconnect these cables.
If you do so, remember to reconnect all before closing the cover!

This chapter describes primarily the connection of universal inputs and outputs, the serial interface and all situations that a technician may face during common servicing.

5.1. Description for GSM gateway

Upper Cover Face
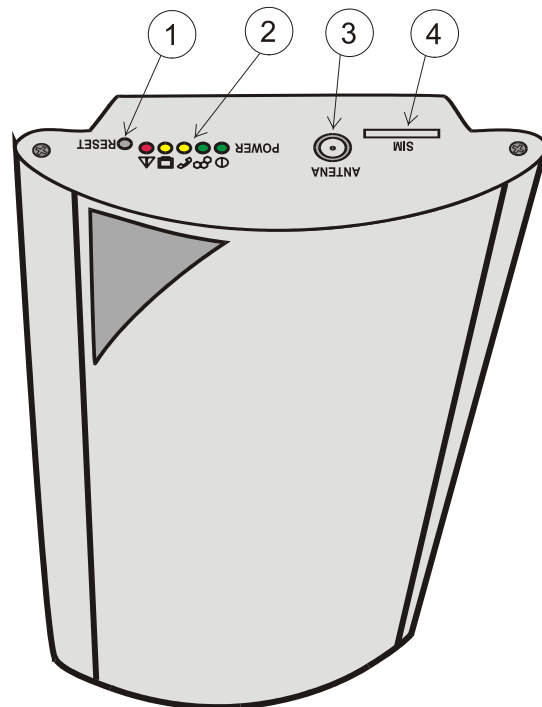
Fig. 7: Upper cover face
1. reset pushbutton
2. LED diodes
3. antenna connector
4. hole for SIM handling

Reset Pushbutton:

The button can be pushed using a tool such as a match, pencil, etc. and has the following functions:

- By pushing the button once during the GSM Gateway operation you reset the equipment. The program is terminated and restarted. This function has no influence on the GSM Gateway set-up stored in the GSM Gateway memory.
- By keeping the button pushed during the GSM Gateway power-on you enter a special mode where you can load a new software version into the GSM Gateway. For details refer to the "*Control Software Upgrade*" chapter.

Antenna Connector:

On models 501061E and 501063E, this connector is not earthed! While the GSM Gateway metal cover is connected with the protective socket wire and thus earthed (as Security Regulations require), the GSM Gateway electronic circuits (on these models) are not earthed. This is advantageous when a PC is connected to the GSM Gateway: by connecting a PC to the RS-232C serial interface (see later) that is earthed to another ground potential (another mains circuit), you earth the GSM Gateway electronic equipment through this PC and data transmission is not interfered by a ground potential difference. **In that case, you need no opto-coupler isolation of the serial port even if the PC is tens of metres distant.** This, of course, is possible only if the antenna connector does not get in touch with the GSM Gateway cover or the earthing thereof to another ground potential.

On models 501100E and 501105E, whole electronics including antenna connector is earthed (connected to PE pin of AC plug and to all parts of metallic cover).

SIM Card Holder:

To insert or replace your SIM card remove the upper cover face with a tool (crosshead screwdriver No.1). This gives your SIM card a better protection against misappropriation.

5.1.1. Bottom Cover Face

Telephone Line Connectors:

This model has two RJ-12 connectors: left one for PBX or phone, right one for PSTN line. The telephone lines are connected to the central pair of pins (two pins nearest to the connector axis). The polarity is arbitrary. The electric isolation of the PBX and GSM Gateway is located as follows:

- Always in PBX,
- During PSTN call, GSM Gateway is isolated from PSTN line, which is physically interconnected to PBX.

Mains Supply Connector:

The mains supply connector is used for PCs and is thus identical in practically all countries. In all countries, a power cable is used whose other end (wall socket end) meets local regulations and socket dimensions. The protective pin (in the middle) is connected with the GSM Gateway cover and used as the first over voltage protection stage for the telephone line circuits. For security and functional reasons, it is necessary that the pins earthed!
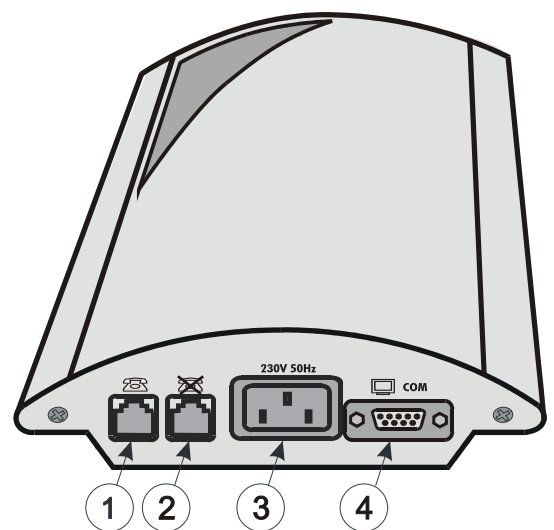
Fig. 8.: Bottom cover face:
1 - PBX line connector
2 – PSTN line connector
3 – AC mains connector
4 – RS-232C serial interface connector

RS-232C Serial Interface Connector:

Since the GSM Gateway in its data mode is a regular modem, the connector pins are exactly the same as in an ordinary modem, see Fig. 9. For the PC connection, a non-cross-over (1:1) extension cable – the same as for the connection of a PC and external modem – is used. The maximum cable length is in excess of 30 metres and depends on the PC – it may be a little trial and error is needed to find an exact maximum length.

Fig. 9.: Serial RS-232C interface connector

Pin	Signal
1	DCD
2	RxD
3	TxD
4	DTR
5	GND
6	DSR
7	RTS
8	CTS
9	RI
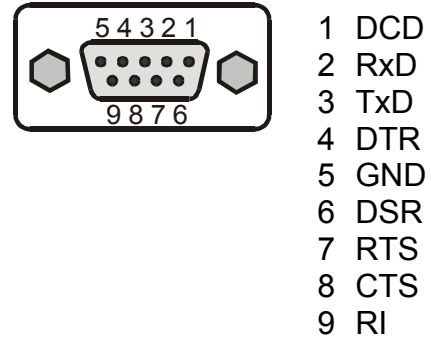
5.1.2. Front Cover Part Disassembly

The remaining elements such as the fuse or the input and output terminals are not accessible until you remove the front cover part, which is clicked into the bottom cover part slots and held by its own flexibility. First remove the upper cover face to grasp the front cover part easily and pull it out. The front cover part includes a panel with LED indicators – the connection cable is detachable from the motherboard.

Fig. 10: Front cover disassembly
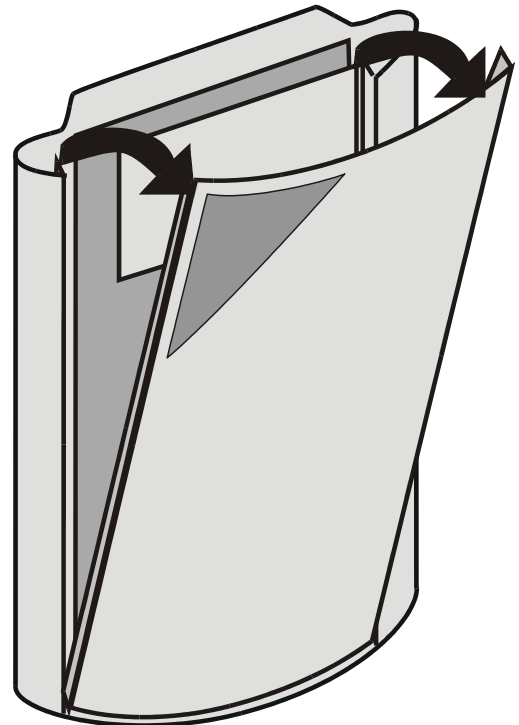
5.2. Description of GSM Gateway PCB
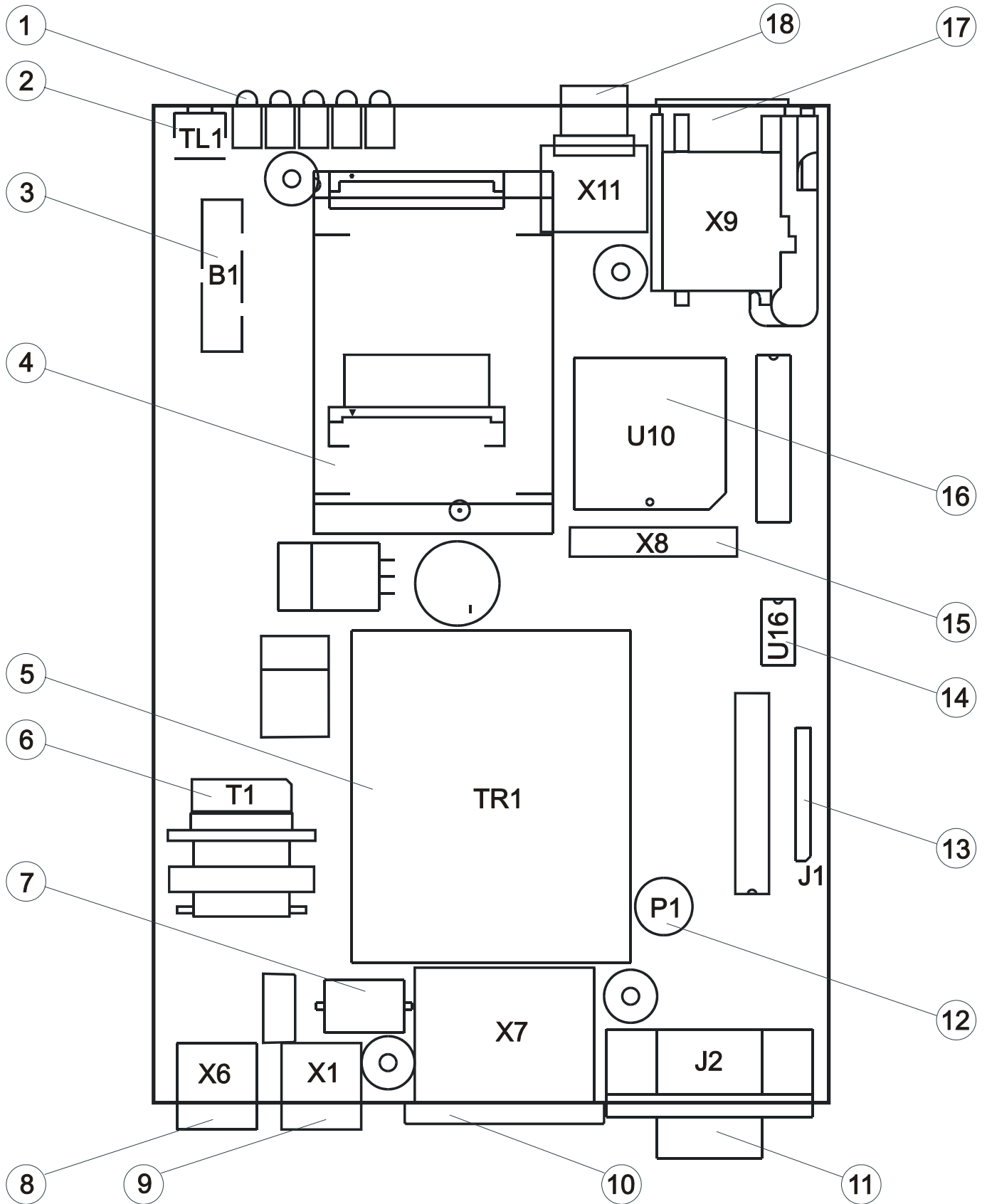
Explanatory Notes

1. LED indicators
2. TL1 - Reset pushbutton
3. Lithium battery in holder
4. SIEMENS $^®$ GSM module TC35 or MC35
5. Mains transformer
6. PSTN line transformer
7. BJ1 - 2 x 10,000 A surge arrester – PSTN line first stage overvoltage protection
8. X6 – PBX line RJ-12 connector
9. X1 – PSTN line RJ-12 connector
10. X7 – mains supply connector
11. J2 – RS-232C serial interface connector
12. P1 - Mains fuse – T 200 m A
13. JP1 - diagnostic connector of power part
14. U16 – EEPROM containing GSM Gateway programmed parameters
15. X8 - diagnostic connector of digital part
16. U10 – main micro controller in the socket
17. X9 – SIM card holder
18. Antenna connector

Notes:

- *The main microcontroller can be removed with a specialized tool only. Usually, it is not necessary because the microcomputer can be reprogrammed in the GSM Gateway. Using another tool may cause damage to or destroy the PCB!*
- *The main microcontroller contains a serial number of GSM gateway as well as a protected code. If erased by a programming tool unlike GSM program, it will not work and these data cannot be re-programmed by GSM program again!*

5.3. Fuse Exchange

General rules:
- Use only a fuse of the same value and type.
- Disconnect the AC power cable while replacing the fuse.
- Fuse for AC power can be replaced only by service which is able to check such parameters as power consumption, DC voltages etc.
- If a fuse fails again, manufacturer must repair equipment.

5.4. Lithium Battery Exchange

	ATTENTION! Explosion risk when the lithium battery is replaced incorrectly. Replace only by the same or equivalent type according the producer's recommendation. Handle the used batteries according to the producer's instructions.

On-board lithium battery is used for RTC (Real Time Clock) during AC main failure only. It is not necessary for GSM gateway operation. A totally discharged battery may have such effects as wrong time information (which is displayed after running GSM program in right top corner of window). Lifetime of battery is more than three years. After this time, a battery change is recommended, or its check (min. voltage 2.9V) at least.

To replace lithium battery, disconnect GSM gateway from mains, open its cover (see chapter 5.1.2.), then remove an old battery using a proper tool and insert a new one.

Battery type: CR2032.

	DANGER! **Do not use metallic tools to handle both old and new lithium batteries during replacement! Don't short it anyway!** **Risk of explosion!**

Recycle or dispose of old batteries in accordance with law and local regulations!

6. Programming

6.1. How to Program

You can program your GSM Gateway in three ways: with a telephone, PC, or remote by PC as listed in the table below:

	Programming method:	Phone	PC	PC, remote
Programming:	Parameters with exception of I2CR table and SMS texts:	✓*)	✓	✓
	I2CR table and SMS texts:	✓	✓	✓
Reading of all parameters:			✓	✓
Upgrade of GSM Gateway firmware:			✓	

*) By reason of extended Call Sorting Table, only records 500 to 599 of this table are programmable by phone. Records A01 to A98 and B01 to B54 are programmable only by PC.

6.2. Before Programming

- Using the chapter 2.7, make sure, that the GSM Gateway works.
- Learn the default set-up of programmable parameters. Keep them as they are unless you need to change them.
- Decide in which way you will program the GSM Gateway. If you can use a PC, then use it. Select the most convenient of the initialising files. Open it with the appropriate program, study the notes therein and change those parameters only that you are not satisfied with.
- If you use the telephone programming, fill all values to be changed into the parameter table fields (chapter 7).
- If the GSM Gateway is not brand-new, make sure that you know the correct service password. If you are not quite sure, perform the full initialisation!

6.3. Handset-Based Programming

6.3.1. Requirements and Recommendations

- You need another extension of the same PBX and a tone-dialling telephone set for programming.
- Use the telephone-based programming only if you do not want to set up many parameters. Remember that you do not have any feedback with a telephone!
- Complete the prepared form first – think before programming!

6.3.2. Entering Programming Mode

- Pick up the handset. If you have a phone connected directly with GSM gateway, go to next step. Otherwise, dial the number to access GSM Gateway. *)
- Wait until you can hear the dial tone ⎯ ⎯⎯ . You cannot program during the current call, or if the GSM Gateway requires a PIN or PUK code.
- To get into the programming mode enter the service password and the ⊠ character. The GSM Gateway transmits a confirmation ♪♪.
- If you make a mistake while entering the password, cancel the entering by hanging-up (the # character cannot be used) and start again.
- The service password is '12345' by default. We recommend changing the password to protect your equipment against unauthorized persons. If you forget the password, your data will not get lost, but you will have to contact the manufacturer.
- If you enter an incorrect password that is not bared by sorting table, the GSM Gateway dials the number and may get through somewhere. This, however, is almost improbable. In case of GSM gateway model for external line and pulse dialling selected, there is no such a risk; GSM gateway sends busy tone if any other number than correct password is entered.

Example: *correct password* ①②③④⑤⊠, *you enter incorrectly* ①①②③④⑤⊠ , *the number is dialled and you will get through to the emergency line 112.*

*) **Note:** You must access GSM gateway by its PBX line interface. Programming by GSM connection (incoming call) or by PSTN call is not possible.

6.3.3. Telephone-Based Programming

Once you have entered the programming mode, you can change all parameters or more parameters in any sequence, except for SMS message texts or I2CR table parameters. It is easy – just enter the parameter number and then the parameter value. Use the ⊠ as separator ("Enter").

Each parameter number has three to four digits (refer to the *Parameter Tables*). After the number and ⊠ are entered, the GSM Gateway sends a confirmation ♪♪ signal if such parameter exists and is available in the particular model and software version. If it is not available, the GSM Gateway transmits a rejection signal ♪♪♪♪♪. After the value and another ⊠ are entered, the GSM Gateway sends a storing ♪♪♪ signal if the value lies in the allowed interval. If not, the GSM Gateway transmits a rejection signal ♪♪♪♪♪. The programmed values are stored immediately during the storing signal ♪♪♪ transmission.

Programming Example:

⑼⓪⑵⊛⑴⑵⑶⓪⓪⓪⊛

Sets the GSM Gateway clock at 12 o'clock, 30 minutes, 0 seconds.

> ⚠ **Important warning!!!** While setting time parameters note the units they are set in – seconds or milliseconds. Since you cannot enter the decimal point from your telephone, milliseconds must be used wherever seconds are too rough. However, you cannot enter any number in the allowed interval (173 ms, e.g.), but you have to respect the **step** prescribed for the particular parameter. If one step is 100 ms, e.g., your value must end with two zeroes.

Example:
Parameter 243, tariff pulse length, has a step of 10 ms. Hence you cannot enter 66 ms, e.g., but 70 ms. To set 1 second, you have to enter 1000.

6.3.4. Programming of Multi-Parameter Line Tables

Some parameters constitute a two-dimension table (the Call Sorting Table, e.g.). In the table, one line has a three-digit number and can contain up to 10 parameters. The individual parameters on the line have a four-digit number, differing in the last digit. They can be entered as follows:

- **Each parameter separately:** follow the preceding procedure, but use a four-digit parameter number;
- **More parameters in sequence:** this method is quicker and is as follows:

1) First enter a three-digit number of the parameter line.
2) After the number and ⊛ are entered, the GSM Gateway sends a confirmation signal ♪♪ if such parameter line exists and is available in the particular model and software version. If it is not available, the GSM Gateway transmits a rejection signal ♪♪♪♪♪.
3) Then enter the first parameter and ⊛. The GSM Gateway transmits a storing signal 2 ♪♪♪♪ if the parameter is in the allowed interval. This signal is different from the common storing signal ♪♪♪ (refer to the *Signalling During Programming Table*) and indicates that the value has been stored and the following parameter is awaited.
4) The programmed value is stored immediately during transmission of the storing signal 2 ♪♪♪ or ♪♪♪♪.
5) Then, you can enter the next parameter and ⊛. The procedure is the same.
6) To signal that this was the last parameter to be programmed, press ⊛ and #.
7) The remaining parameters on the line will not be changed.
8) If you enter all parameters, the GSM Gateway transmits a common storing signal ♪♪♪ after the last one.
9) Therefore, you cannot pass onto the next line by simply entering parameters; you have to enter the line number.
10) To program a table line in the above-described way, you always have to start with the first parameter.

Example:

⑸⓪⑴⊛⓪⑹⓪⑵⊛⑴⊛⓪⊛#

Stores the first 3 parameters into line 501 (to be specific, the 0602 prefix is allowed and the call will be routed to GSM).

6.3.5. Programming Error

- If you make a mistake while entering a number (no matter whether a parameter number or a value) and find it before you press ⊠, you can cancel the whole programming step using the 🔲 character.

- If the GSM Gateway transmits a rejection signal ♪♪♪♪♪, you have to re-enter the parameter number even if the value was incorrect.

- If, while programming more parameters at the same time (table line) using the above mentioned procedure, you make a mistake in the third parameter, e.g., and the GSM Gateway transmits a rejection signal ♪♪♪♪♪, all correctly entered parameters have already been stored and you need not re-enter them. You can either re-enter the whole line or enter the remaining parameters individually.

- If you program and the GSM Gateway stores a value that you do not want, you can re-enter the value correctly, of course, i.e. re-enter the parameter number, etc.

6.3.6. Erasing of parameters

- To erase parameter, enter its number, ⊠ and once again ⊠.
- Only parameters, which can be empty (see Parameter Tables, chapter 7), can be erased this way. In case of parameter, which cannot be empty (defined range, e.g. receiving volume and the like), GSM Gateway transmits a rejection signal ♪♪♪♪♪.
- Erasing parameters in the Call Sorting Table: If you will erase some prefix, i.e. the first parameter on the row, all parameters on this row will be erased (or initialised). If you will enter row number (e.g. 501) or number of first parameter (e.g. 5011), it has the same effect. But, if you will enter number of another parameter (e.g. 5010), only this parameter will be erased.

6.3.7. Programming End

Hang up to terminate programming. After the programming end make sure that the GSM Gateway works as you want it to. Store the completed programming form of configuration file safe.

6.3.8. Signalling During Telephone-Based Programming

Signal	Name	Meaning
♪♪	Confirmation	• After valid service password • After valid parameter No.
♪♪♪♪♪	Rejection (error)	• After invalid parameter No. • After invalid parameter value • After cancelling of anything by 🔲
♪♪♪	Storing	• Value is valid and stored
♪♪♪♪	Storing 2	• Value is valid and stored, next parameter may be entered

6.4. PC - Based Programming via serial interface

6.4.1. Connecting to PC, starting of GSM program

Connect GSM gateway and your PC or notebook by RS-232C Serial Interface. Use an enclosed serial cable or another common 1:1 male/female serial cable. Run GSM program (actual version is available on internet). Program works without connected GSM gateway too (prearrangement of set-up, demo...). It has three basic functions, described below.

6.4.2. Programming tool

Basic function of GSM program is to view and edit all parameters of GSM gateway. Parameters are grouped in windows, which are according with parameter tables in chapter 7 of this manual. Help and hints are available for all parameters, as well as a choice of basic set-up. You can add your own notes and save all settings as a file for future use. Parameters can be up- or downloaded, but some are read-only or write-only; e.g. passwords cannot be read, and serial number cannot be overwritten.

6.4.3. Upgrade of GSM gateway firmware

GSM gateway is equipped with ISP (In System Programming) feature. You can "upgrade" it whenever you wish. Actual version of firmware is available on Internet (free).

How to upgrade your GSM gateway firmware:
1) Download actual version of GSM program and Hex file for upgrade from www.2n.cz
2) Install GSM program on your PC, connect GSM gateway to serial port
3) Run GSM program and select used COM
4) Check connection - GSM program must be able to read settings from GSM gateway
5) **Turn GSM gateway OFF** (in a case of model with battery back-up, disconnect battery too)
6) **Press RESTET button and hold it** while turning GSM gateway ON
7) Most lamps on GSM gateway must light up - upgrade mode is activated
8) **Make an upgrade** now - button **Hex** in the top toolbar of GSM program window.

6.5. Remote Programming by PC

6.5.1. Remote Supervision Purpose

Remote supervision allows reading and changing the GSM gateway configuration remotely, thus saving time of servicing technicians enabling them to solve some problems remotely. For remote supervision, the same GSM software is used as for local setting using a serial interface. All features are absolutely identical with the exception of the GSM gateway firmware upgrade, which **cannot be done** remotely.

6.5.2. What You Need to Run Remote Supervision

On client's premises:
1. The GSM gateway must be equipped with a **SIM card supporting data transmission**.
2. The GSM gateway must contain firmware of version 2.36 and up.
3. Parameters required for remote supervision must be selected in the GSM gateway.

In Supervision Centre:
1. You need to know the client's GSM gateway service password.
2. A GSM program of version 1.05 or up must be installed in the PC.
3. The PC must be equipped with a reliable modem (analog, ISDN, or GSM), or a GSM gateway made by 2N, which works in the data mode. If the PC is equipped with an analog modem, it has to be connected to a CLIP-sending exchange (calling line identification presentation).
4. The ‚incognito' function **may not** be activated on the Supervision Centre side.

Important warning!

Once programmed for remote supervision by selected line, client's GSM gateway cannot receive an ordinary call from this line. The GSM gateway with the said selection regards every call containing the Supervision Centre CLIP as a remote supervision attempt. Hence, to make such a call, you should use another line or another SIM card, or the ‚incognito' function, or make a call in direction from client's GSM gateway to the Supervision Centre.

Important recommendations
1. Select a reliable telephone line or SIM card for connection whose number will not be changed in the future (will not be renumbered).
2. You are recommended to use another GSM gateway or a GSM modem rather than a fixed line modem to establish connection much more quickly.
3. The GSM gateway or GSM modem in the Supervision Centre should be equipped with a SIM card with a separate number for data connections (necessary if you use remote supervision with call-back to a GSM gateway of different operator).
4. A connection within one GSM network is more reliable than a connection between different operators.
5. If you are forced to use two different operators for remote supervision, read the "Data Connection Problems" chapter carefully.

6.5.3. What Is "Call-back"

In case of normal supervision call, the calling party, i.e. the Supervision Centre covers data transmission costs. To avoid billing complications, the GSM gateway can establish remote supervision connection on the account of the client's GSM gateway. It is the so-called call-back:

1. The Supervision Centre GSM gateway or modem "calls" the client's GSM gateway.
2. The called GSM gateway identifies the Supervision Centre CLIP as a call-back request and rejects the incoming call immediately. Therefore, this request is free.
3. Right after that, the called GSM gateway calls the Supervision Centre number that sent the request. If unsuccessful, it retries several times later.
4. The Supervision Centre GSM gateway or modem waits for this call-back. Having received it, it establishes data connection. Remote supervision is possible upon entering of the valid service password.

6.5.4. Remote Supervision from 2N Supervision Centre

You need not configure anything for this service; all you have to do is to set parameter 930 to enable remote supervision. Necessary parameters are hidden (cannot be programmed). Call-back is always used for this connection.

6.5.5. Overview of Remote Supervision Parameters

One common parameter enables / disables remote supervision, and four sets of parameters follows:

- CLIP - number of remote supervision Centre,
- Call-back enabled / disabled,
- Number for call-back.

These four sets enables to make remote supervision call from up to four different sites. Parameters are described in details in chapter 7.6 - *Service parameters*.

6.5.6. How to make GSM Gateway ready for Remote Supervision
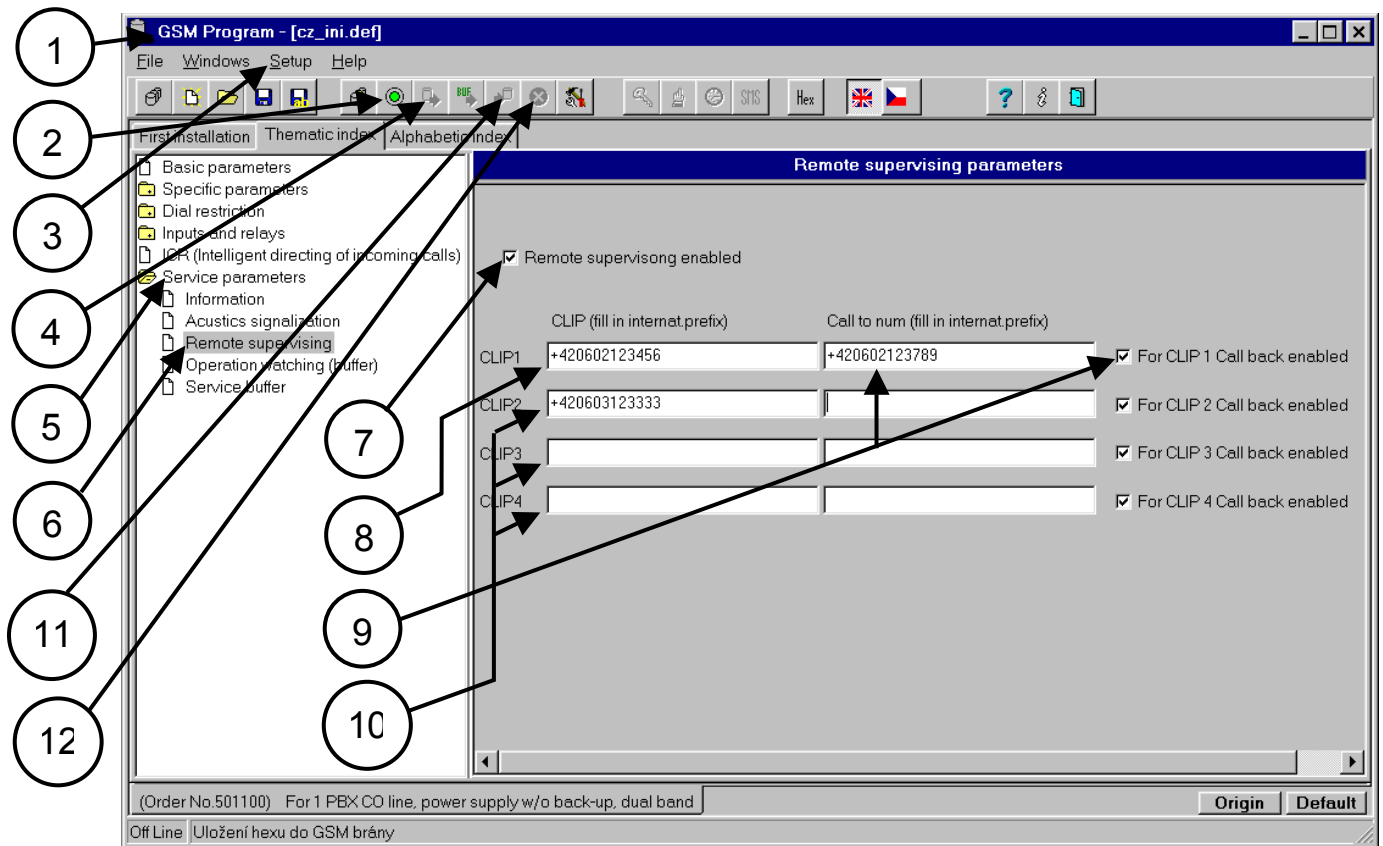
1. Connect the GSM gateway to your PC and run the GSM program.

2. Switch into the on-line mode.

3. If the connection fails, select the correct serial port number in the 'Setting' menu.

4. Read the GSM gateway setup.

5. Open the 'Service Parameters' item.

6. Open the remote supervision parameter folder.

7. Make sure that remote supervision is enabled. If not, enable it.

8. Enter the number of the modem or GSM gateway, which is used for remote supervision.

9. If you select the call-back item, and the GSM gateway, which is used for remote supervision, has a separate data number, enter the data number in the right-hand column.

10. You can enter up to 4 numbers to be used for remote supervision.

11. Remember to save the setup back into the GSM gateway.

12. Switch into the off-line status.

Important recommendation:

You are recommended to make this setup before installing the GSM gateway at the client's. With older types of GSM gateways already installed, you are advised to make this setup immediately after software upgrade to a version supporting remote supervision – 2.36 and up.

6.5.7. How to Run Remote Supervision
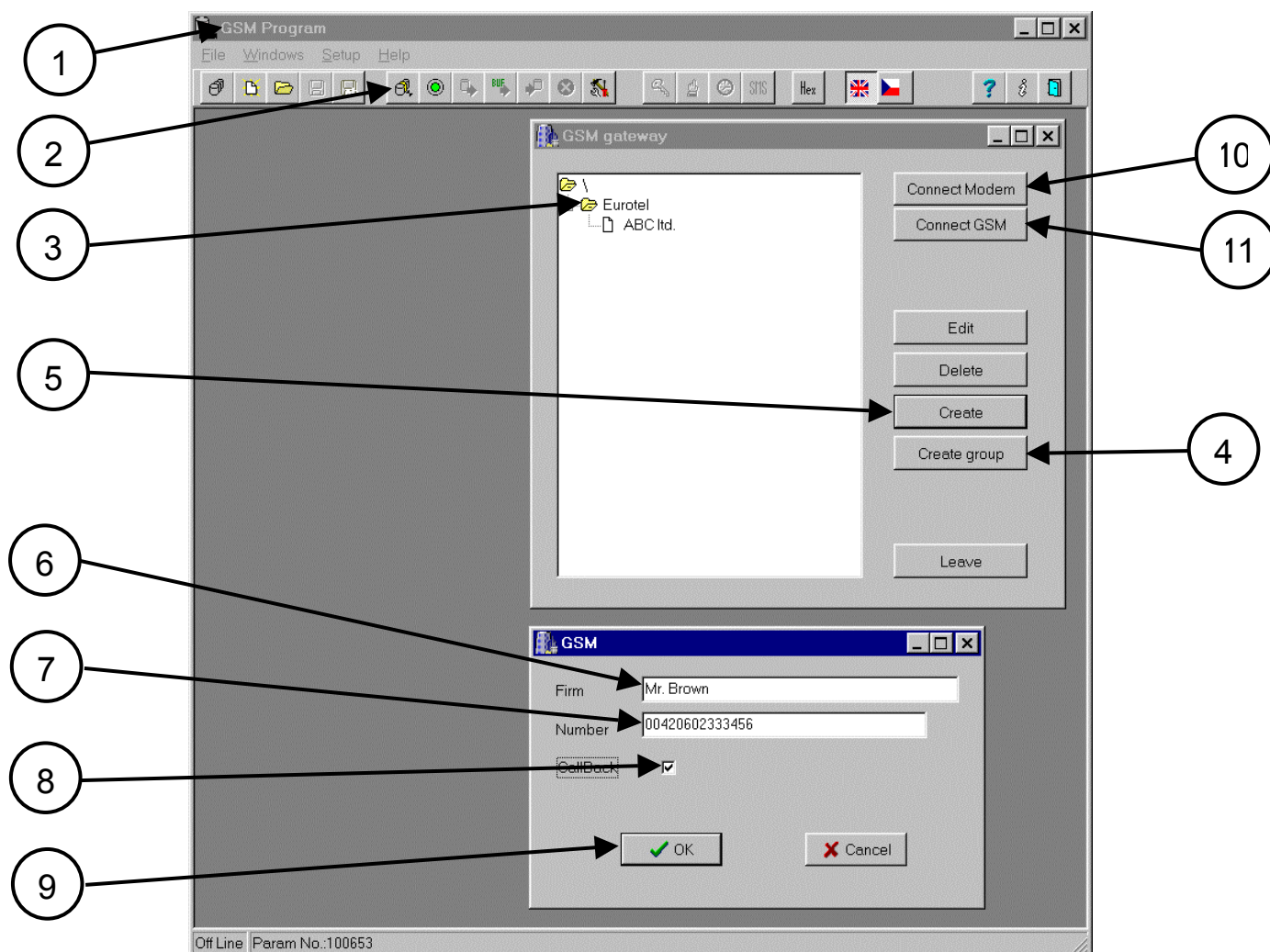
1. Run the GSM program on the Remote Supervision Centre PC.
2. Click on this icon to open the list of supervised GSM gateways.
3. The list may include subfolders in multiple levels.
4. Click on this key to create a subfolder.
5. Click on this key to create a new item in the list.
6. Enter client's name, company etc. here.
7. Enter the respective GSM gateway telephone number here.
8. **WARNING!** Settings here must correspond with the respective GSM gateway setup in order that the remote supervision connection can be established successfully!
9. Click on this key to save the record in the list.
10. Click on this key to establish remote supervision through your modem.
11. Click on this key to establish remoter supervision through your GSM gateway.

Important cautions:

- Connection establishing is the quickest without call-back or with a GSM – GSM connection made within one and the same GSM network, which usually takes only a few seconds. With call-back and different GSM operators, the establishing may take up to several minutes.
- With call-back, the PC does not check whether the call-back comes from the GSM gateway you sent the request to. Hence, if you, e.g., try to establish connection with one GSM gateway and cancel the attempt before getting through, and then you try to make connection with another GSM gateway, you should make sure from which of the gateways the call-back is coming!

6.5.8. Remote Supervision Establishing – Flow Chart

```
┌──────────────────────────┐
│        Remote            │
│  Supervision PC calls    │
│  client's GSM gateway.   │
└──────────────────────────┘
              │
              ▼
┌──────────────────────────┐
│   Client's GSM gateway   │
│      gets a RING.        │
└──────────────────────────┘
              │
              ▼
         ╱ Is it ╲
  YES ◄──◄ Incognito calling ╲
         ╲  (CLIR)? ╱
              │ NO
              ▼
         ╱ Is CLIP ╲
  NO ◄───◄ the Remote ╲
         ╲ Supervision Center ╱
          ╲  number? ╱
              │ YES
              ▼
         ╱ Is ╲
         ╱ callback ╲
         ╲ activated for this ╲──► YES
          ╲  CLIP? ╱
              │ NO
              ▼
         ╱ Is the ╲                    ┌──────────────────────────┐
VOICE ◄──◄ incoming call ╲             │ GSM gateway rejects the  │
         ╲ voice or data ╱             │         call.            │
          ╲ connection? ╱              └──────────────────────────┘
              │ DATA                              │
              ▼                                   ▼
   ┌──────────────┐          ┌──────────────────────────┐
   │    Answer.   │          │  GSM gateway waits for 5 │
   └──────────────┘          │        seconds.          │
              │              └──────────────────────────┘
              ▼                              │
   ┌──────────────────────┐                  ▼
   │ Connection           │            ╱ Is there ╲
   │ established.         │            ╱ a callback ╲──► YES
   └──────────────────────┘           ╲ number for the ╱
              │                        ╲ received CLIP? ╱
              │                             │ NO
              ▼                             ▼
```

Rejection

End.

GSM gateway dials back the same number.

GSM gateway dials the callback number.

Has data connection been established? YES / NO

Remote Supervision Center asks for service password entering.

Have 5 callback attempts been made? YES / NO

Password entered, remote supervision started.

GSM gateway waits for 60 seconds..

6.5.9. Data Connection Problems

GSM data transmission is different from analog CO line data transmission: it is necessary to specify in advance whether the case will be voice, data or fax connection. With the connection once established, the connection mode cannot be changed. Therefore, the calling party should select the required connection before establishing it and this information should get to the target GSM network operator through all networks involved. Experience from other countries and operators show that this is quite impossible in some cases and not fully reliable in other cases. For an overview of remote supervision options with/without call-back see the table below.

Service	Client's GSM Gateway	No Call-back	With Call-back
GSM gateway	One and the same GSM operator	No problems	No problems, you need not select the data number for call-back.
GSM gateway	Another GSM operator, SIM **has no** special data number.	No guarantee; it works if both operators are able to establish data connection between each other, in the direction from service to client.	No problems if the call-back data number is used.
GSM gateway	Another GSM operator, SIM **has** a special data number.	No problems if the called GSM gateway data number is used.	No problems if the call-back data number is used.
ISDN modem	Any GSM operator	No guarantee; it works only if both operators are able to establish data connection between each other, in the direction from service to client.	No guarantee; it works only if both operators are able to establish data connection between each other, in the direction from client to service.
Analog modem *)	Any GSM operator, SIM **has** a data number.	No problems if the called GSM gateway data number is used.	No problems
Analog modem *)	Any GSM operator, SIM **has no** data number.	Not applicable	No problems

*) **IMPORTANT!** The analog modem must be connected to a CLIP-sending PBX.

Table – SIM Card Identification Options for Outgoing Calls:

Voice number	Data number	Number used for outgoing call identification:
YES	NO	voice number
YES	YES	voice number
NO	YES	data number

Note: The SIM card can have another number for fax calls, which is no important for remote supervision.

7. Parameter Tables

7.1. Basic Parameters

Parameter		Par. No.	Min.	Max.	Def.	Your choice	Remarks
PIN entering mode		101	0	1	1		0 = manually, 1 = automatically
PIN – value		102	4 dig.	8 dig.	-		*Write-only parameter!*
CLIR – incognito		103	0	2	0		0 = by operator setting 1 = no CLIR, 2 = send CLIR
Volume - direction to GSM		104	1	5	3		1 = minimum, 5 = maximum
Volume - direction from GSM		105	1	5	3		1 = minimum, 5 = maximum
Roaming enable/disable		106	0	1	0		0 = disabled, 1 = enabled
Incoming calls rejection		107	0	1	1		0 = reject all incoming calls, 1 = normal operation
GSM signal level indication		108	0	1	0		0 = disabled, 1 = enabled
Delete SMS when SIM is full		109	0	1	1		0 = disabled, 1 = enabled
Silence after dialling		111	0	25500	0	e.g. 10000	Time [ms], step is 100. Recommended value = 10000
End of dialling signal length		112	0	2550	200		Beep length [ms], step 10
Dialling end time-out		113	2000	15000	6000		In milliseconds! Step = 100 ms
Switchboard operator number		114	Max. 8 digits				
International prefix		115	1 dig.	4 dig.	00		Usually 00. Equal to "+" on GSM networks.
Country code		116	1 dig.	4 dig.	*)		Sooner parameter 6002. Enter without "00" prefix, "+" will be added automatically.
Long distance code		117	0 dig.	2 dig.	0		It may be empty - in some countries it is not used.
Operator prefix		118	1 dig.	4 dig.	-		Reserved for future use.
Tone after end of call		121	0	2	2		0 = silence, 1 = continuous, 2 = busy tone ─ ─ ─ ─ ─
Busy tone	1st tone	122	0	2550	320		Step = 10 [ms]
	1st pause	123	0	2550	320		
Dialling tone	1st tone	131	0	2550	320		Step = 10 [ms]
	1st pause	132	0	2550	320		
	2nd tone	133	0	2550	640		
	2nd pause	134	0	2550	640		
	3rd tone	135	0	2550	0		
	3rd pause	136	0	2550	0		
Dialling tone during roaming	1st tone	141	10	2550	320		Step = 10 [ms]
	1st pause	142	10	2550	320		
	2nd tone	143	0	2550	320		
	2nd pause	144	0	2550	320		
	3rd tone	145	0	2550	640		
	3rd pause	146	0	2550	640		
DTMF receiver timing	Min. burst	151	30	200	40		Step = 10 [ms]
	Min. pause	152	30	200	40		
DTMF transmitter timing	Tone	153	30	1000	100		Step = 10 [ms]
	Pause	154	30	1000	100		
Receiving DTMF during a call		155	**0**	**1**	**1**		0 = disabled, 1 = enabled

Parameter	Par. No.	Min.	Max.	Def.	Your choice	Remarks
Number for automatic dialling	156	-	18 dig.	-		Empty = automatic dialling OFF
Time-out for automatic dialling	157	0	15	7		[sec], 0 = instantly after pick up
Maximal time of outgoing call to GSM	158	0	2550	0		time [s], step = 10 s. 0 = without limit
Maximal time of incoming call from GSM	159	0	2550	0		time [s], step = 10 s. 0 = without limit
SMS centre number	160	-	20 dig.	-		Necessary for SMS reports
COM – Data mode	161	**0**	**1**	**1**		0 = disabled, 1 = enabled
COM – SMS mode	162	**0**	**1**	**1**		0 = disabled, 1 = enabled
Enable sending call information via COM	166	**0**	**1**	**0**		0 = disabled, 1 = enabled
Automatic data calls answering	181	**0**	**15**	**0**		Number of rings before automatic answer. 0-do not answer automatically

Notes to Basic Parameters:

- **Parameters connected with SIM-card protection**

    The "PIN" parameter is used only if "Enter Automatically" is selected and the SIM card requires the PIN. If you enable the SIM-card function without PIN entering, you will not need this parameter. If "Enter Automatically" is selected and you do not enter the PIN, enter a wrong PIN, or replace the SIM card without entering the new PIN, the GSM Gateway will try to enter the old PIN once, identify its invalidity, and invite you, if you try to dial, to enter the PIN manually (by PIN tone ᐨᐨᐨᐨ ᐨᐨᐨᐨ   ᐨᐨᐨᐨ). If you enter the new PIN correctly, it will be saved without programming. If you select the "Enter PIN Manually" mode, you will have to enter the PIN upon every power-on and the GSM Gateway will not store it anywhere.

- **Parameters connected with GSM network**

    CLIR (number of the calling extension, i.e. of your GSM Gateway) is normally sent to the called party. The "Incognito" option should be applied for with your GSM operator (usually for a single fee). WARNING!!! If you have not activated this service with your GSM operator and set the "Incognito" function at 1 = Do Not Send Number, no outgoing calls will be established in some GSM networks!!!

    There is a risk with roaming in frontier areas that, due to a failure in your native GSM network, the GSM Gateway might get registered in a GSM network of the neighbouring country and all GSM calls will be much more expensive. If you enable roaming, you should set your SIM card properly and activate the service with your operator! If you disable roaming here, it is an unambiguous barring command that depends on nothing else.

    Incoming call enable: The GSM Gateway is designed for outgoing calls primarily, because you mostly do not pay for incoming calls and thus cannot cut your telephone costs. If, however, incoming calls occupy the GSM Gateway for a considerable period of time, this might make your outgoing calls more difficult and diminish the expected cost cuts. Should your GSM Gateway be too busy, you are advised to bar incoming calls. To cut incoming call costs too (from employees moving outside the company, e.g.), you can install another GSM Gateway for incoming calls only using a SIM card with pre-paid services (no monthly lump sum will be paid).

- **Volume and echocanceller setting**

    Parameters 104 and 105 allow changing of volume by +/- 6 dB around its nominal value in 3dB steps.

- **Silence after dialling**

  This parameter has no effect. It is used only for GSM gateway with TC35 modules up to firmware version 3.0.

- **Parameters for dialling supervision and intelligent incoming call routing**

  Dialling end identification: If the GSM Gateway does not identify the end of the dialled number by any faster method (refer to the Call Sorting table), it waits for a pre-set timeout. Then (or immediately if it identifies the dialling end by a faster method), it sends a short beep, whose length can be adjusted, and starts establishing the connection with the called party.

- **Parameters of all tones transmitted by GSM Gateway to line**

  These parameters can change basic GSM Gateway tones that are heard by the connected extension. It is mainly the busy tone − − − − − and dial tone − ──. The course of these tones may be important for such automatic devices as the answering and recording machine or telephone set with automatic Redial. If roaming is activated, the special roaming dial tone − − ── is used to warn the subscriber that the connection is more expensive. The pre-set tones meet the requirements of European standards. Tones coming from the GSM network, of course, cannot be changed and may differ from the pre-set tones.

- **Tone dialling parameters**

  The GSM Gateway both receives and transmits tone dialling. Time parameters for reception of tone dialling must be 20 ms shorter at least than the fastest tone dialling to be received by the GSM Gateway (some telephones transmit 70 ms pulses and 70 ms spaces). Time parameters for tone dialling (sending) are used for outgoing calls only if the GSM Gateway dials the extension number.

  The pre-set parameters meet the requirements of European standards.

- **Maximum time of call parameters**

  Parameters 158 and 159 are suitable to set maximum time of GSM call. Range is from 10 sec. to 2550 sec. (42.5 min.) or without limit - default value. 30 sec. before end you can hear warning tone. During last 10s short tone repeats every second. 0-no limit. Parameter 158 is applicable for outgoing calls, parameter 159 for incoming calls. PSTN calls cannot be limited by this parameters.

- **Parameters enabling data functions**

  These parameters enable or disable the use of serial port as a modem for sending and reading of SMS on a PC, execution of SMS commands for switches, and transmitting of SMS on input statuses and "Security Centre" function. Each mode of use can be disabled individually. For call supervision, information about all calls including pseudo-tariff metering can be send out via serial interface – se parameter 166.

- **Parameter 181 function**

  This parameter is set to zero by default, it means that the GSM gateway does not connect incoming data calls automatically. By selecting a value other than zero, you activate the automatic data call answering function – the GSM gateway receives the call automatically and, if the data connection is established successfully, sends the "CONNECT" message to the serial interface. In this mode, you can also receive data calls manually by entering the ATA command.

- **Help for parameters:**

  101 = the parameter is only used if the SIM card requires the PIN.

  102 = is used for automatic entering.

  103 = Can disable your number presentation to the called party. Does not work with PSTN.

  104 = Volume control option by +/- 6 dB to the GSM network.

105 = Volume control option by +/- 6 dB from the GSM network.

106 = Disables roaming in frontier areas to prevent the GSM Gateway from registering, in the event of a fault in the own GSM network, in a GSM network of the neighbouring country.

107 = Disables all incoming calls.

108 = Displays the GSM signal intensity every 10 seconds.

109 = If SIM memory is full, oldest SMS will be erased when new one will come in – to enable receiving commands for switches.

110 = A greater value can reduce the echo, but it can cause some distortion of speech – only for GSM gateways with module SIEMENS M20.

111 = Value tells, how long silence will be after end of dialling. Very long value can "cut" begin of ringing tone, but not begin of call.112 = Sets the length of the beep announcing the dialling end.

113 = Timeout for the GSM Gateway to wait for further dialling.

114 = Extension number to ring upon an incoming call, except for intelligent routing. On a PBX C.O. line used only if DISA is available in the PBX.

115 = This parameter is used for CLIP conversion within Intelligent Incoming call routing and programming by phone (entering phone numbers).

116 = It is used for Intelligent Incoming call routing for CLIP conversion. This parameter is accessible also as parameter 6002 in section Intelligent Incoming Call Routing.

117 = It is used for Intelligent Incoming call routing for CLIP conversion.

118 = Reserved for future use.

121 = If a remote extension hangs up earlier the extension connected to the GSM Gateway can hear the tone selected here.

122-123 = Using these two parameters you can change the busy tone rate.

131-136 = Using these six parameters you can change the dial tone course. It is a tone signalling that the GSM Gateway is ready for dialling.

141-146 = Using these six parameters you can change the roaming dial tone course. This tone should be different from the normal dial tone to signal that the connection is much more expensive. If roaming is disabled, these parameters are not used.

151-152 = Using these two parameters you can change the tone dialling receiver features. Higher values may make the receiver not receive all dialled digits!

153-154 = these parameters can accelerate the transmission of tone dialling. They are only used for incoming calls if the GSM Gateway dials the extension number.

155 = receiving of DTMF during a call is necessary for function "write to I2CR table".

156 = Number for automatic dialling ("baby-call). Automatic dialling is OFF as long as this parameter is empty.

157 = Time-out for user to start dialling, in case of present number for automatic dialling. If this time is over and user is not dialling any number, GSM gateway starts automatic dialling.

158 = Time after that outgoing call to GSM is automatically interrupted. 30s before end you can hear warning tone. During last 10s short tone repeats every second. 0-no limit.

159 = Time after that incoming call from GSM is automatically interrupted. 30s before end you can hear warning tone. During last 10s short tone repeats every second. 0-no limit.

160 = the number necessary for the sending of any SMS.

161 = this parameter can disable or enable the data mode, i.e. the use of the serial port as a modem.

162 = this parameter can disable or enable the use of the serial port for SMS sending and reading on a PC.

165 = Using this parameter you can disable or enable the "Security Centre" function.

166 = This parameter enables sending out information about calls via COM

181 = This parameter defines after how many rings an incoming data call will be answered automatically. If set at zero, the function is disabled.

7.2. PBX's CO Line Interface Parameters

Parameter			Par. No.	Min.	Max.	Def.	Your choice	Remarks
Received dialling (DTMF / pulse)			201	0	1	0		0=DTMF, 1=pulse
Time parameters for pulse dialling receiver	Make min.,		204	-	-	20	-	Time parameters are **fixed**: make 20 to 80 ms, break 30 to 90 ms.
	Break min.,		205	-	-	30	-	
Flash min.			206	10	2550	100		
On-hook min.			207	10	2550	500		
Time parameters for ringing	1st ring pulse		211	0	25500	1000		Step = 100 [ms]
	1st pause		212	0	25500	4000		
	2nd ring pulse		213	0	25500	0		
	2nd pause		214	0	25500	0		
	3rd ring pulse		215	0	25500	0		
	3rd pause		216	0	25500	0		
Max time of ringing			224	0	255	0		[s], 0 = unlimited
Max off-hook time without action	Busy tone		225	1	255	30		Step = 1 [s]
	Power down		226	0	255	60		0 = power down disabled
Signalisation of begin of GSM connection	Type		231	0	5	0		See notes below
	Time		232	10	2000	100		Step = 10 [ms]
Signalisation of end of GSM connection	Type		233	0	5	0		See notes below
	Time		234	10	2000	100		Step = 10 [ms]
Frequency of signalisation pulse for begin or end of connection			235	0	9	6		See notes below
Pseudo-tariff pulses transmit or no			241	0	1	0		0 = no, 1 = yes
Level of pulse transmitter			242	0	1	1		0 = low, 1 = high
Length of tariff pulse			243	10	1000	100		Step = 10 [ms]
Min. pause between tariff pulses			244	10	1000	200		
Frequency of tariff pulses			245	0	9	6		See notes below
DISA dialling start time			251	100	25500	4000		Step = 100, [ms]
Dial switchboard operator number in the course of DISA			252	0	1	0		0=no, 1=yes

Notes:

231 and 233: Signalling of connection start or end:

0	None signalling
1	Current break for time specified by par. 232 or 234
2	Polarity change (without return) Not recommended.
3	Polarity change with return, time of inverted polarity specified by parameter 232 or 234
4	Frequency impulse (typically 16 kHz), length see parameter 232 or 234, frequency see parameter 235
5	Polarity change (without return, always): parameter 231 - polarity is changed on the beginning of **both** incoming and outgoing calls, parameter 234 - polarity is changed on the end of call even if PBX hangs up first.

NOTE: **Only for GSM calls! During PSTN calls, signalling is not generated by GSM gateway, it is only passed thru from PSTN, if present.**

235 and 245: Frequency of tariff pulses or signalisation pulse for begin and end of connection:

2	12 kHz
6	16 kHz
Other	Reserved

Notes to PBX <u>CO</u> Line Interface Parameters:

It is very **<u>important</u>** not to interchange dialling **<u>receive</u>** and **<u>send</u>** parameters! Dialling send parameters meet the applicable standards; while dialling receive parameters must be set with a sufficient reserve to receive even considerably damaged dialling. This applies generally to both tone and pulse dialling modes.

The maximum timeout for a passively seized line is a pair of parameters determining the time for delay in dialling after picking up the line, or delay in hang-up after hang-up by the other subscriber. Parameter 226 is usually set at a higher time value than parameter 225. Then, the GSM Gateway first transmits the busy tone and then, if you do not hang up, disconnects the line current. After that, it tests the line every minute until the line is hung-up.

Parameters 231 – 235 – Dialling start and end signalling are used only if the PBX requires this signalling, for call cost billing, e.g.

Parameters 241 – 245 – Pseudo-tariff rating is used only in case the PBX requires this signalling or if a coin-operated device is connected. It works during GSM calls. Tariff pulses during PSTN calls must be transmitted by public telephone exchanges.

Help for parameters:

201 = Selects the dialling type to be received by the GSM Gateway from the PBX or a telephone set. Gateway receives only dialling of selected type; different dialling is ignored!

204-205 = Using this parameter you can change the pulse dialling reception features. Higher values may make the receiver not receive all digits dialled!

206 = Minimum line current interruption time that is identified as "Flash" by the GSM Gateway. The parameter is not used yet.

207 = Minimum line current interruption time to be identified as "Hang-Up" by the GSM Gateway.

211-216 = Using these six parameters you can change the ringing course. Use 2, 4, or 6 parameters as needed, leaving zeroes for the remaining ones. (The selected course is repeated.)

224 = Maximum ringing time – but it is also limited by the GSM network (30 seconds, e.g.).

225 = If you do not start dialling after picking up the line, or do not hang up after call termination, the GSM Gateway will start sending the busy tone after this timeout.

226 = If you do not start dialling after picking up the line, or do not hang up after call termination, the GSM Gateway will disconnect the line current after this timeout.

231 = Method of informing of the PBX by the GSM Gateway that the connection has been established.

232 = Signal time used for informing of the PBX by the GSM Gateway that the connection has been established.

233 = Method of informing of the PBX by the GSM Gateway that the connection has been terminated.

234 = Signal time used for informing of the PBX by the GSM Gateway that the connection has been terminated.

235 = The signal frequency time used for informing of the PBX by the GSM Gateway that the connection has been established or terminated – if frequency signalling is selected.

241 = Using this parameter you can disable or enable the transmission of tariff pulses during GSM calls to be counted by the GSM Gateway according to the Call Sorting table data.

242 = Tariff pulse intensity selection – use more intensive signals unless they disturb calls.

243-244 = Using these two parameters you should select the highest tariff pulse transmission rate.

245 = Using this parameter you can select the tariff pulse frequency.

251 = Time after seizure, when PBX is able to receive CO line tone dialling (DISA).

252 = By selecting NO you allow the calling subscriber (except for intelligent routing) to hear the PBX DISA message and to dial the extension itself.

7.3. Call Sorting Table

7.3.1. Purpose

This table describes GSM Gateway's behaviour (during outgoing calls) somehow depends on the called number. Usually, it is not difficult to recognize, according to a few first digits (the prefix), an international call, mobile network call, special service call, emergency call, etc. It is possible to enter a line into this table for each such group of numbers to define:

- **Call routing – PSTN or GSM**
- Whether this number can be dialled or is barred (call restriction);
- How to detect when the dialling is complete;
- How to bill the call (the so-called pseudo tariff metering);
- Whether and how the called number is to be modified before being sent to the GSM or PSTN network;

These "rules" can be written into the table, for more details see below.

7.3.2. Sorting Principle

Every table line contains one prefix (of variable length) plus parameters that describe the behaviour of the GSM Gateway, if the beginning of the called number is identical to this prefix. However, there is often an exception to the rule – a number that starts identically but has to be served in a different way.

*Example 1: Trunk calls usually start with **0**, but international calls with **00**. It is possible to define a trunk call by one of the following three ways:*

1. *The trunk call starts with **01, 02, 03, 04, 05, 06, 07, 08,** or **09** – very demanding.*
2. *The trunk call starts with **0[1...9]** – a more convenient way.*
3. *The trunk call starts with **0**, except for **00**, which is an international call.*

The table uses the last of the three ways cited above. There may be any number of exceptions and the description is very easy: <u>**any table line that starts with the same prefix but includes one or more additional digits is considered an exception.**</u>

Example 2: To bar all international calls <u>except for</u> calls to the Czech and Slovak Republics, complete 3 table lines – in any sequence – as follows:

00	*– international calls*	*– barred*
00420	*– Czech Republic*	*– allowed*
00421	*– Slovak Republic*	*– allowed*

The said setting has the following advantages:

- Any further line can be added any time without modifying or checking the remaining ones. The new line becomes automatically „an exception" to any of the preceding ones if it starts with the same prefix.
- There may be more exceptions to one line and there may be an exception to an exception again, etc.

Example 3:

0	*– trunk calls*	*– allowed, routed to **PSTN***
00	*– international calls*	*– barred*
00420	*– Czech Republic*	*– allowed, routed to **PSTN***
004206	*– Czech Republic*	*– allowed, routed to **GSM***

In this example, there are 3 levels of exceptions, and <u>last row shows a prefix for GSM routed calls.</u> Of course, there are even more complicated situations in reality. Nevertheless, the table is still quite comprehensible.

7.3.3. Table Structure and Parameter Ranges

Par. No.	xxx1 Begin of number	xxx2 Call enable	xxx3 call routing	xxx4 Number length	xxx5 End #	xxx6 End *	xxx7 Initial tariff	xxx8 Impulses per [ms]	xxx9 Take away	xxx0 Append
501	Max. 16 digits 0...9, *, #	0 = no 1=yes	0=GSM 1=PSTN	0,3...16	0 = no 1=yes	0 = no 1=yes	0... ...255	0 or 100 to 999900	0 – 16	Max. 16 digits 0...9, *, #
⋮	⋮	⋮	⋮	⋮	⋮	⋮	⋮	⋮	⋮	⋮
598	-//-	-//-	-//-	-//-	-//-	-//-	-//-	-//-	-//-	-//-
A01	-//-	-//-	-//-	-//-	-//-	-//-	-//-	-//-	-//-	-//-
⋮	⋮	⋮	⋮	⋮	⋮	⋮	⋮	⋮	⋮	⋮
A98	-//-	-//-	-//-	-//-	-//-	-//-	-//-	-//-	-//-	-//-
B01	-//-	-//-	-//-	-//-	-//-	-//-	-//-	-//-	-//-	-//-
⋮	⋮	⋮	⋮	⋮	⋮	⋮	⋮	⋮	⋮	⋮
B54	-//-	-//-	-//-	-//-	-//-	-//-	-//-	-//-	-//-	-//-
599	**Other**	-//-	-//-	-//-	-//-	-//-	-//-	-//-	-//-	-//-

NOTE: Records A01 to A98 and B01 to B54 are programmable only by PC.

Values after Initialisation:

Par. No.	xxx1 Begin of number	xxx2 Call enable	xxx3 call routing	xxx4 Number length	xxx5 End #	xxx6 End *	xxx7 Initial tariff	xxx8 Impulses per [ms]	xxx9 Take away	xxx0 Append
501 to 530	Empty	1=yes	0=GSM	0	1=yes	0 = no	0	0	0	Empty

The table also includes a special line (500), where setting a few parameters can disable the whole table or some columns. It is a quick way to the „emergency operation" if there is an error in the table and you have no time to locate it.

Line 500 – Table Blocking Parameters

Par. No.	5001	5002	5003	5004	5005	5006	5007	5008	5009
Meaning	General enable	Call enable	General routing	Number length	End #	End *	Initial tariff	Impulses per [ms]	Take off and append
Value range	0 = no 1 = use table	0 = no 1 = use table 2 = all	0 = GSM 1 = use table 2 = PSTN	0 = no 1 = use table	0 = no 1 = use table 2 = yes	0 = no 1 = use table 2 = yes	0 = no 1 = use table	0 = no 1 = use table	0 = no 1 = use table
Default value	1	1	1	1	1	1	1	1	1

General enable: if you disable the whole table with a zero, all numbers will be governed by line 599 – „Others".

Remarks and Explanatory Notes to Parameters

- If you program the GSM Gateway using a PC, on each line you can add a comment that is saved but not sent to the GSM Gateway.

- The table has now 250 records; rows *A01 to A98 and B01 to B54 are programmable only by PC.*

- Since the table is shared by several functions, the sorting must be detailed enough to cover all purposes: if, for example, pseudo tariff metering is used, calls with different cost have to be distinguished even if all of them are allowed.

1 – Number Beginning: used for call type identification (GSM, trunk, free, etc.). The „Number Beginning" field in line 599 cannot be filled - the parameters on this line are used automatically for all numbers, which are not found in table.

2 – Call Barring: the parameter says whether the dialled number is allowed or barred.

3 –The most important parameter in this model; tells, which way the call will be routed.
Note: Today's version of software sends busy tone always, when requested network is inaccessible. Future version will have more fine settings - GSM only, PSTN only, GSM prior to PSTN, and vice versa.

4 – Number Length: The parameter defines how long the number can be expected to be for the given prefix. Therefore, the dialling into GSM network can start as soon as the required number of digits is dialled. 0 means do not observe the number length – enter zero here, if the number length is not fixed. If, however, the maximum number length is known at least, you can enter it too.

5, 6 – Dialling End with '*' or '#': These characters can be a part of dialling with some prefixes (GSM services, e.g.). Here they <u>cannot</u> be used for terminating the dialling. With other prefixes (fixed PSTN, e.g.) they are not used as a part of dialling, and here they can be used for terminating the dialling. The GSM Gateway can then start dialling as soon as the ‚*' or ‚#' character appears in the number.

Note: If '' or '#' or the number length cannot be used for terminating the dialling, then a timeout will be used (programmable).*

7 – Starting Tariff: Represents the minimum call cost. It is the number of tariff pulses that are sent upon the call connection **additionally** (the pulses are transmitted closely one after another). Works only with calls to GSM network.

8 – Tariff Pulse: It is a time between two pulses – hence, the more expensive the call, the lower the value. It is set in milliseconds to be as precise as possible – one step is 100 ms. a zero means do not send tariff pulses. **Works only with calls to GSM network.**

Note: It is safer to complete tariff rates with barred numbers too in case you enable these numbers later.

9 – „Remove" and 0 – „Add": These parameters help forward calls automatically to the Internet Call, e.g., or translate emergency numbers (PSTN to GSM), etc. The „Remove" item says how many characters are to be removed from the number beginning, the „Add" item contains the characters to be added – the number of these characters can be different, of course. For example, you remove nothing, and add ‚42', or remove ‚150' and add ‚112', etc. Moreover, you can use the parameter for your GSM Gateway access protection: add a „password" for selected calls (international, e.g.) and then remove it. Those, who don't know the password, cannot dial this prefix. See row 510 in table 7.3.4.

Help for parameters:

5001 = This parameter can disable searching in the table. In this case, last row "Other" is used for all numbers.

5002 = This parameter can disable an use of a the column "Call enable". In this case, all numbers can be enabled or bared by this parameter; but a fitting row is searched and the other parameters are used.

5003 = This parameter can disable an use of a the column "Call routing". In this case, all calls can be directed to GSM or PSTN. It can be usual e.g. if one network is not working, to redirect all calls to second one.

5004 = This parameter can disable an use of a the column "Number length".

5005 = This parameter can disable an use of a the column End #. In this case, a character # is not interpreted as an end of dialling generally.

5006 = This parameter can disable an use of a the column End *. In this case, a character * is not interpreted as an end of dialling generally.

5007 = This parameter can disable a transmitting of initial tariff pulses on the beginning of a GSM call.

5008 = This parameter can disable a transmitting of tariff pulses during a GSM call.

5009 = This parameter can disable an use of columns "Take away" a "Add".

5991 = This parameter not exists! Row 599x is used for numbers, which are not matching with any row in the table.

5992 = This parameter enables or disables all numbers, which are not matching with any row in the table.

5993 = This parameter tells, if calls are routed to GSM or PSTN. It is used for all numbers, which are not matching with any row in the table.

5994 = This parameter can define a maximum length of all numbers, which are not matching with any row in the table.

5995 = This parameter defines, if a character # is interpreted as an end of dialling for all numbers, which are not matching with any row in the table.

5996 = This parameter defines, if a character * is interpreted as an end of dialling for all numbers, which are not matching with any row in the table.

5997 = This parameter can set a number of initial tariff pulses on the beginning of a GSM call. It is used for all numbers, which are not matching with any row in the table.

5998 = This parameter can set a period of tariff pulses during a GSM call. It is used for all numbers, which are not matching with any row in the table.

5999 = This parameter can remove a selected number of digits from the beginning of number. It is used for all numbers, which are not matching with any row in the table.

5990 = This parameter can contain a number to be added to the beginning of number. It is used for all numbers, which are not matching with any row in the table.

7.3.4. Example of a Table

Note: I'm sorry this example (GSM Gateway T- Mobile, location Czech republic, Prague) is not very helpful for other countries; better one is under preparation.

Par. No.	xxx1 Begin of number	xxx2 Call enable	xxx3 call routing	xxx4 Number length	xxx5 End #	xxx6 End *	xxx7 Initial tariff	xxx8 Impulses per [ms]	xxx9 Take away	xxx0 Appe nd	Notes
501x	0	yes	PSTN	-	yes	yes			0	-	Trunk calls
502x	02	no	PSTN	-	yes	yes			0	-	Prague
503x	020	yes	PSTN	-	yes	yes			0	-	020 is not Prague!
504x	072	yes	GSM	10			1	60	0	-	GSM Eurotel
505x	073	yes	GSM	10			1	120	0	-	GSM T-Mobile
506x	077	yes	GSM	10			1	60	0	-	GSM Oscar
507x	0609	no		-			1	1,5	0	-	„VAT services"
508x	0800	yes	PSTN	10					0	-	Free services
509x	00	no	PSTN	-					0	-	International calls
510x	7531	yes	PSTN	-	yes	yes			4	00	International via password
511x	0042072	yes	GSM	14			1	60	0	-	GSM Eurotel
512x	0042073	yes	GSM	14			1	120	0	-	GSM T-Mobile
513x	0042077	yes	GSM	14			1	60	0	-	GSM Oscar
514x	112	yes	GSM	3			0	0**)	0	-	Emergency call
⋮											
530											Empty
599	Other	yes	PSTN	6					0	-	

Notes:

- *As you can see, columns 7 and 8 are not filled for PSTN calls, because tariff impulses can be only passed thru from PSTN.*

- *The above-mentioned tariff rates are fictitious, not corresponding to any real situation!!!*

- *Empty fields are not important for this example.*

7.3.5. Call Sorting Table – Form for Your Needs

	xxx1	xxx2	xxx3	xxx4	xxx5	xxx6	xxx7	xxx8	xxx9	xxx0	
	Begin of number	Call enable	Call routing	Num-ber length	End #	End *	Initial tariff	Impulse [ms]	Take away	App end	Your notes
		0 = no 1=yes	0 = GSM 1=PSTN	0,3... 16	0 = no 1=yes	0 = no 1=yes	0 – 255	0 - 999900	0 - 16	*)	*) Max. 16 digits 0...9, *, #
599x	Other										

7.4. Intelligent Incoming Call Routing Table

The Intelligent Incoming Call Routing (refer to the User Manual) allows the GSM Gateway to call a specific extension upon having found the caller's number (CLIP) in the Intelligent Incoming Call Routing table. This saves the switchboard operator's time and the calling party's time and money. This is possible only during calls from GSM network. GSM gateway cannot receive CLIP form PSTN line.

CAUTION: The programming of the Intelligent Incoming Call Routing table cannot be made by phone. This is because data can be entered in the table during a call, which is much more effective (refer to Chapter 3.7.). Within programming by a PC, you can read and display the whole table, find some important lines, move them to the top and then lock this part of the table against overwriting.

Basic Parameters for Intelligent Routing:

Par. No.	Parameter	Range	Def.	Your Value	Note
6001	Intelligent routing enable	0 - 1	1		0=no, 1=yes
6002	Your international prefix	1-4 digits	-		enter without "00", "+" is added
6003	Table recording password	3-8 digits	777		No password deletion by initialisation!
6004	Number of locked records	0 - 99	0		Locked area starts at 601x

Related Parameters – Refer to Chapters 7.1. to 7.2.

Operator number	114	
International prefix (usually 00 - default)	115	
Your international prefix (*accessible also as 6002*)	116	**Note:** If operator number is not entered and dialling for incoming call is not enabled, all incoming calls with unknown CLIP are rejected.
Long distance code (usually 0 - default)	117	
DISA beginning timeout	251	
Dial operator number with DISA	252	
Enable dialling for incoming call	310	
Ring to switchboard operator after timeout	351	
Ringing timeout for switching to switchboard operator	352	

CAUTION! Remember to enter the above-mentioned related parameters correctly to ensure a proper function of intelligent routing!

Intelligent Routing Table Structure:

Meaning	CLIP	Caller's Name	Extension Number	Extension Name
Range	Max. 20 digits. * or #	Max. 32 characters	Max. 8 digits	Max. 32 characters
Par. No.	 1		 2	
601x				
...	...		...	
649x				

Notes:
- *The whole table (601x through 649x) is blank and unlocked after initialisation.*
- *CLIP numbers are entered including international prefixes, starting with +, not 00.*
- *If 0 (zero) is entered, calling party with this CLIP can dial any extension (by DTMF).*
- *The text fields for "Caller's Name" and "Extension Name" are available on the PC only; they are not saved in the GSM gateway but in a file. They are used for a better orientation in the table.*

Help for parameters:

6001 = Enables the Intelligent Incoming Call Routing according to the caller's number if included in the Intelligent Routing table.

6002 = your own international prefix is used for the completion of the called number to be included in the Intelligent Routing table.

6003 = Defines the password for the "Forward this caller to extension ... next time" command.

6004 = Locked records (starting from the table beginning) are protected against overwriting and have priority.

7.5. Parameters for "Dial-thru" model

Parameter	Par. No.	Min.	Max.	Def.	Your choice	Notes
Max. off-hook time for PSTN line in case of no dialling	701	1	255	10		*) [s]
Min. on-hook time for PSTN line	702	100	25500	1000		*) [ms]
Max. pause in ringing from PSTN	703	100	25500	4500		**)
DTMF burst time	704	100	25500	100		***)
DTMF break time	705	100	25500	100		***)
Time between off-hook and dialling to PSTN	706	100	25500	1000		***)

Notes:

) These two parameters are used during an outgoing call. PSTN line is already off hook (since PBX line off-hook). When dialling is finished, GSM gateway checks how long is PSTN line off hook. If parameter 701 is over, GSM gateway hangs up for a moment, using parameter 702.

**) It is recommended to measure ringing waveform from PSTN and set this parameter exactly. Otherwise, following situations may occur:*

   a) ***Parameter is too short:** during ringing from PSTN, after each ring this time-out will expire earlier than next ring will start. In this "window", if someone will access GSM gateway from PBX to make an outgoing call, these two people may be connected (and calling party from PSTN will hear DTMF dialling) or calling party from PSTN will hear a silence and then connection will be terminated.*

   b) ***Parameter is too long**: after unsuccessful call from PSTN, if someone will access GSM gateway from PBX to make an outgoing call just at the moment when calling party from PSTN hanged up, GSM gateway will consider this call as an incoming one. Call will be always made via PSTN.*

According to our experience, both instances are very rare. First one may be more unpleasant, second one can only decrease savings (very little). Therefore it is better to set parameter 703 little bit longer than necessary (10 seconds default value is big enough).

***) Parameters 704 - 706 are used when dialling is repeated to PSTN. Pulse dialling is not possible yet.*

Help to parameters

701, 702 = In a future upgrade, perhaps these parameters will be used to protect PSTN line against busy tone, power down or failure state which can be caused by long off-hook during GSM call.

703 = Time between two rings from PSTN. It is not recommended to set less than 6 seconds.

704 = Time of DTMF digit transmitted to PSTN

705 = Time between two DTMF digits transmitted to PSTN

706 = Time between Off-hook and start of DTMF dialling

7.6. Service Parameters

Parameter		Par. No.	Range, format	Def.	Your choice	Notes
Service password		901	3 – 8 digits	12345		Write only parameter!
Time		902	hhmmss			
Date		903	ddmmyy			
HW version		911	0-255			Read only parameter!
SW version		912	1.0 – 9.99			Read only parameter!
Serial No.		913				Read only parameter!
GSM module serial No.		914				Read only parameter!
Password for upgrade No. I.		921				
Password - Dial Restriction only		922				Write only parameter!
Password - lock to operator.		923				
Remote supervision enable		930	0 / 1	1		Applies to 2N Supervision Centre too.
CLIP – Supervision Centre number *)	number 1	9311	max. 18 dig.	-		Dial the number including international prefix, e.g. +420261301111 **)
	number 2	9321	max. 18 dig.	-		
	number 3	9331	max. 18 dig.	-		
	number 4	9341	max. 18 dig.	-		
Call-back enable / disable	for number 1	9312	0 / 1	1		0 = off 1 = on
	for number 2	9322	0 / 1	1		
	for number 3	9332	0 / 1	1		
	for number 4	9342	0 / 1	1		
Call-back numbers ***)	for number 1	9313	max. 18 dig.	-		
	for number 2	9323	max. 18 dig.	-		
	for number 3	9333	max. 18 dig.	-		
	for number 4	9343	max. 18 dig.	-		

*) It is the main SIM card number to identify the calling SIM card. It is because the SIM card can have one or more (up to three) numbers. In case of calls incoming from an analog network, the GSM operator differentiates, according to the required number, which service is required by calling party - voice, data or fax connection. With outgoing calls, these numbers are not needed because only one is used for identification.

**) If entered by phone, enter number including international prefix too - e.g. 00420261301111. In some countries pay attention to parameter 115, international prefix may be different than 00.

***) If different than numbers 1 to 4, i.e. data numbers, if exists

Help to Parameters:

930 = this parameter can disable remote supervision including the 2N Supervision Centre.
9311, 9321, 9331, 9341 = Number to identify the calling Supervision Centre. If the Supervision Centre SIM card has two numbers (voice and data), enter the voice number.
9312, 9322, 9332, 9342 = These parameters selects individually for each number, if call-back will be used or not.
9313, 9323, 9333, 9343 = Number to be called in case the call-back function is on. If it is identical with the number requesting call-back, you need not fill this parameter. If the Supervision Centre SIM card has two numbers (voice and data), enter the data number.

Commands for initialising:	No.	Parameter
Basic parameters	991	Valid service password must be entered as a parameter as verification to avoid inadvertent initialisation e.g. when this function is entered by mistake. Default value of service password is 12345.
Interface for external line of PBX	992	
Interface for local line of PBX	993	
Universal inputs and outputs	994	
Call sorting table	995	
Intelligent Incoming Call Routing Table	996	
Parameters for "Dial-thru" model	997	
Operation monitoring and signalling	998	
Complete initialisation	999	

Notes:
901 – Service Password: *If you change the password, remember it well. If you forget it, contact the GSM Gateway manufacturer. The service password is not changed by initialisation!*

Warning:

GSM gateways, shipped to some customers, may contain pre-programmed values for use in target country, to make installation as easy as possible. Commands 995, 996 and 999 erases these settings, i.e. local GSM operator prefixes in call sorting table and international prefix - parameter 6002.

8. Miscellaneous

8.1. Telephone Cost Saving Tips

- **Fill in Call Sorting Table carefully; keep it actual when operators come up with new prefixes.** Remember that this model of gateway is a routing device.

- Decide, depending on your call traffic, whether to purchase a more GSM Gateways. Best way is to use GSM gateways for all PSTN lines of your PBX.

- Select the appropriate GSM tariff rate while purchasing your SIM card. Since a high outgoing load is expected, select higher lump-sum rates that mostly offer lower fees per called time unit.

- If your employees are equipped with mobile telephones, they will be able to call your company more cheaply via the GSM network than through the fixed PSTN.

- To make the best of your GSM Gateway for outgoing calls, you can bar incoming calls or keep your GSM Gateway number secret.

- Sometimes it is convenient to install a separate GSM Gateway for incoming calls. Here you just need a SIM card with pre-paid credit and you need not pay a monthly lump sum. Incoming calls then do not block your "main" favourable-rate GSM Gateway.

- If your PBX allows so, set up the automatic seizure of a mobile network by the *ATEUS* ® - DIAL-THRU GSM GATEWAY to maximize your savings.

- With advanced PBXs, it is possible to set up call barring statuses for each user separately. To maximize your GSM telephone cost economy divide the users into the following three groups:

   1. No calls into the GSM network (busy tone − − − − − when the user tries to call).
   2. Calls into the GSM network only via the GSM Gateway (busy tone − − − − − when the GSM Gateway is busy).
   3. Calls into the GSM network also via PSTN lines if the GSM Gateway is busy.

- You can bar selected telephone numbers directly on your GSM Gateway.

- Find how many "free minutes" and what types of calls are provided by the selected tariff rate. If, for example, the rate provides free minutes into the fixed network, you may lose these minutes.

All PBX programming options described herein are offered by *ATEUS* ® PBX systems made by 2N TELEKOMUNIKACE a.s..

8.2. Trouble Shooting

If, after the GSM Gateway's power-on, all LED's are on (except for the 1st and 4th from the upper end), any the following situations has occurred:

- If the GSM Gateway does not beep, you have probably pressed the RESET button during the power-on and the GSM Gateway is waiting for software reprogramming.
- If the GSM Gateway gives a permanent tone, some inapplicable software has been loaded (for a different GSM Gateway model).
- If the GSM Gateway beeps intermittently, there is an EEPROM error.

If you forget the service password, remove the EEPROM, erase it in a programmer and it push back into the socket**.** The default values, including the service password, will be reset. Retrieval of stored settings is possible but it is necessary to contact manufacturer.

8.3. List of abbreviations

- **CLIP** - Calling Line Identification Presentation
- **COM** - serial port of computer
- **DCD** - Data Carrier Detect - signal which indicate established data connection
- **DISA** - PBX's function allowing calling party to enter (by DTMF only) a required extension number
- **DTMF** - Dual Tone Multifrequency - (worldwide standard of dialling)
- **EEPROM** - Electrically Erasable Programmable Read Only Memory - Memory independent of power supply (non-volatile memory)
- **FXO** - line interface dedicated to ordinary phone devices (worldwide standard)
- **FXS** - line interface electrically equivalent with ordinary phone (complementary to FXO)
- **GND** - Ground - contact or line connected with ground, or common reference point of whole electronic circuit, which may be connected with ground
- **GSM** - Group Switched Mobile system - today's digital mobile cellular phone network standard, used in Europe another parts of word, excluding USA and Japan
- **HW** - Hardware - in this context we mean any electronic equipment, circuit, board, component etc.
- **IN** - Input - input terminal or contact
- **I2CR** - Intelligent Incoming Call Routing
- **LCR** - Least Cost Routing - software function which finds a cheapest path to make required connection
- **OUT** - Output - output terminal or contact
- **PBX** or **PABX** - Personal (Automatic) Branch Exchange
- **PC** - Personal Computer - computer compatible with IBM PC standard
- **PIN** - Personal Identification Number - a short password, used to protect SIM against unauthorized usage; after a few wrong attempts SIM card block itself automatically
- **PUK** - Personal Unblocking Key - long password, used to unblock SIM if blocked (see PIN) and enter a new PIN value
- **RS-232C** - worldwide standard for serial port
- **SIM** - Subscriber Identity Module - module with chip, which is a carrier of information about subscribers phone number in GSM network
- **SMS** - Short Message Service - more often it is used for this message itself
- **SSR** - Solid State Relay - semiconductor switch, used like a relay
- **SW** - Software (any program)
- **TTL** - Transistor-Transistor Logic - standard for digital circuits, definition of voltages for levels 0 a 1
- **PSTN** - Public Switched Telephone Network
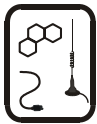
9. Technical Parameters

GSM:

GSM network type	GSM phase II
SIM card	plug-in 3 V, „small"
Transmitter power	2 W / 900 MHz 1W / 1800 MHz
Receiver sensitivity	-104 dBm
GPRS class (model 501105)	class 8

Antenna:

Frequency	900 / 1800 MHz
Impedance	50 Ω
Power	2W
Antenna connector type	FME (male)
Cable length	3 to10 m or without cable

Mains:

230V AC, tolerance:	230 V$\pm$10%, 50 or 60 Hz
Power consumption	Max. 15 VA
Mains connector	Euro connector (PC)
Lithium battery type	CR2032

PBX line interface

Interface type	2-wire, FXO
Connector type	RJ-12, 6/2
Off-hook AC impedance	600 Ω
On-hook line voltage	24 V DC
Off-hook line current	Max. 30 mA
Line loop resistance	Max. 800 Ω
Dialling tone frequency	425 Hz
Dialling type	Tone (DTMF) or pulse
Ringing voltage	45 V_{RMS}, 50 Hz
Surge current	2x 250 A (8/20 μs)

PSTN line interface

Interface type	2-wire, FXS
Connector type	RJ-12, 6/2
Off-hook AC impedance	600 Ω
On-hook quiescent current	Max. 100µA @ 60V
Off-hook line current	10 - 65 mA
Off-hook line voltage	typ. 4V @ 30 mA
Dialling type	Tone (DTMF) only
Ringing voltage	min. 20 V_{RMS}, 25 - 50 Hz
Ringing impedance	4 - 10 kΩ @ 50 Hz
Surge current	2x 10.000 A (8/20 μs)
Inputs - level "0"	Max. + 0.8V DC
Inputs - level "1"	Min. + 2,1V DC
Inputs - maximal voltage	Max. +/- 24V

Serial Interface:

Interface type	RS-232C
Connector	D-Sub 9 pins, female
Interface data rate	19200 bit / sec

Other:

Dimensions (w/o connectors)	150 x 150 x 55 mm
Operating temperature	0°C to 45°C
Air humidity	5 to 95%

Use the product as designed and manufactured and in accordance with these instructions for use.

The manufacturer reserves the right to modify the product in order to improve its qualities.

The *ATEUS* ® - DIAL-THRU GSM GATEWAY contains no environmentally harmful components. When the product's service life is exhausted and you find no other application for it, dispose of it in accordance with applicable legal regulations.